SIP IP Phone

VIP-155PT User's manual

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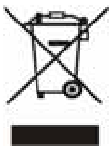
The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning



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Revision

User’s Manual for PLANET SIP IP Phone:

Model: VIP-155PT

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Part No. EM-VIP155PTV1

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Chapter 1

Introduction

Overview

Meeting the next-generation Internet telephony service demands, PLANET Technology provides feature-rich, toll-quality Internet telephony service solutions. The 802.3af Power over Ethernet (PoE) IP Phone —VIP-155PT brings cost-effective solution for voice communications and interoperates VoIP hardware and systems from major third party vendors with traditions of PLANET VoIP family. As a feature-rich IP Phone, the VIP-155PT fulfills your needs. The VIP-155PT is SIP 2.0 (RFC3261) compliant with SIP digest authentication supports. And the VIP-155PT is the cost-effective SIP PoE IP Phone.

The VIP-155PT feature high-quality speakerphone technology; also include an easy-to-use speaker on/off button and call hold/transfer buttons for various voice services. These features go beyond the conventional voice systems nowadays, and the PoE IP phones are cost-effective solution for Internet Telephony Service Provider (ITSPs) communications and interoperate VoIP hardware and systems form othe major third party vendors with the traditions of PLANET VoIP family.

As feature-rich IP Phones, the VIP-155PT fulfill your needs. They are simple to use, and have additional features such as built-in PPPoE/DHCP clients, password-protected machine management, large LCD menu display, hands-free speakerphone, last number redial, incoming message indicator, and user-intuitive web administration system.

The VIP-155PT are self-contained, service-integrated IP phones — offers intelligent phone features, and powerful voice processing power. The VIP-155PT can effortlessly deliver toll voice quality equivalent to the regular PSTN connections utilizing cutting-edge Quality of Service, echo cancellation, comfort noise generation and voice compensation technology. Meanwhile, the dual Ethernet interfaces on the VIP-155PT allow users to install in an existing network location without interfering with desktop PC network connections. The new VIP-155PT deliver more convenience, efficiency, innovation and benefits of VoIP in your dailylife.

VIP-155PT Functions

- **Simple Installation and administration**

Configuration of the VIP-155PT can be performed in minutes via the keypad, or web interfaces. Using the built-in LCD display, the VIP-155PT offers user-friendly configuration guidelines, machine operation status, call status displays, and incoming call identification.

- **Feature-rich keypad IP Phone**

  The VIP-155PT integrates a high-quality speakerphone with the Call Hold, Forward and Transfer functions and also provides advanced telephone features, such as 9 speed-dial keys, last number redial, incoming call history, Auto Answer indicator in a much more convenient and functional manner than traditional telephone sets.

- **Dynamic IP address assignment, and voice communication**

  The VIP-155PT can act as a PPPoE/DHCP client, automatically obtaining an IP address for Internet access.

- **Various field applications compliant**

  The VIP-155PT is capable of handling both peer-to-peer and SIP proxy registration, authentication to interact with major SIP gateway/IP Phone in the market. The VIP-155PT offers the most flexibility and interoperability with PLANET and 3rd party VoIP vendors, allowing the deployment of both simple and complex VoIP networks such as ITSP, PC-to-Phone/Phone-to-PC or enterprise VoIP environments.

- **Standards compliant**

  The VIP-155PT complies with SIP 2.0 (RFC3261), interoperates with 3rd party SIP voice gateways/terminal/software as well as other PLANET VoIP products. Supported Voice codecs and VoIP technologies are: G.723, G.729ab, G.711u-law/a-law; Voice Activity Detection (VAD), and the Confort Noise Generation (CNG).

- **NAT Optimization, Firewall policy packet filtering and QoS mechanism**

  The VIP-155PT provides user definable policy-based firewall protection, and a packet filtering mechanism to prevent business or residential network from malicious attacks or intrusion. The firewall policy offers VoIP administrators access control privilege choices to apply to LAN users to restrict Internet access or prevent improper use.

Package Content

The contents of your product should contain the following items:
IP Phone
Power adapter
Quick Installation Guide
User's Manual CD
RJ-45 cable x 1

Physical Details

The following figure illustrates the front/rear panel of VIP-155PT.

Front View and Keypad function

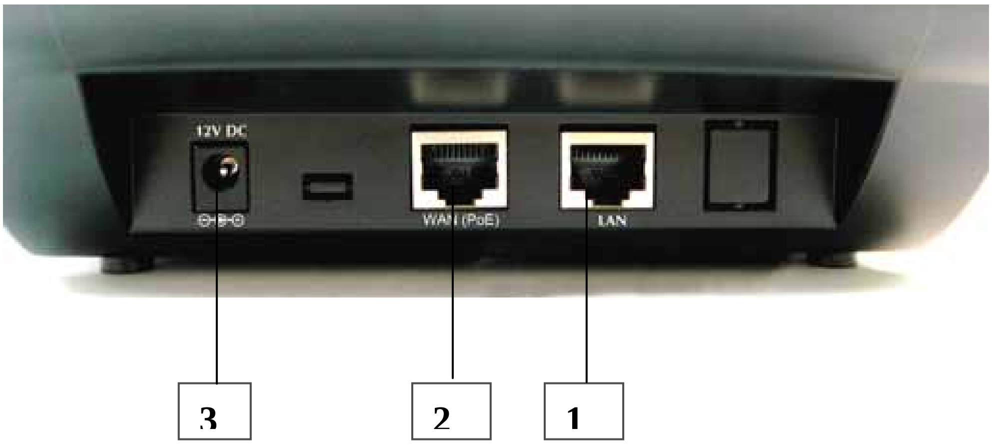
Front Panel of VIP-155PT

Keypad Description

1	**LCD Display**	Menu and all status shall be displayed for users.
2	**Speed Dial No.1~No.9**	To make a speed dial call by pressing the speed dial key No.1 ~ No.9.
3	**Sysinfo**	Circularly show phone number, wan ip, registration status, server ip address, gateway and mask info.
4	**Out call**	Show the outgoing calls history.
5	**FWD**	To transfer an active call (incoming call answered or outgoing call accepted) to another IP phone.
6	**Send**	After complete dial digits, press this button to make call.

7	**Redial**	Press to dial the last dialed number when the IP Phone is off-hooked.
8	**PWR**	The green light goes on when power on.
9	**Message**	The green light goes on-off when there is an incoming call. The light goes constant on when there have voice message (Proxy Mode.)
10	**Handfree**	To switch between the usage of the handset and the speaker devices.
11	**Vol+**	To increase the volume of voice when at off-hooked state. To page up menu when at configuration mode.
12	**Mute**	Press to mute sounds when at talk mode.
13	**Menu/OK**	To bring out the menu selection while IP Phone is in idle state. To be used as confirm configuration or enter sub-menu.
14	**Modify**	Press to modify the configuration.
15	**Exit**	To escape to an upper layer menu selection.
16	**Up**	To increase the volume of voice when at off-hooked state. To page up menu when at configuration mode.
17	**Down**	To decrease the volume of voice when at off-hooked state. To page down menu when at configuration mode.
18	**In call**	Show the incoming calls history.
19	**Pbook**	Enter the phone book selection.
20	**Record**	Enter the Voice Record selection.
21	**Hold**	To hold the conversation.
22	**Vol-**	To decrease the volume of voice when at off-hooked state. To page down menu when at configuration mode.
23	**Del**	Delete digits when at Calling and Configuration modes.

Rear View



Rear Panel of VIP-155PT

1	**LAN**	RJ-45 connector, to maintain the existing network structure, connected directly to the **PC** through **straight** CAT-5 cable
2	**WAN (PoE)**	RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable. Please connect the WAN interface when using IEEE802.3af PoE power supply (**PT** model only)
3	**12V DC**	12V DC Power input outlet

ⓘ Hint

- The Power over Ethernet support on PLANET VIP-155PT complies with the 802.3af standards. Using non-802.3af compliant PoE device will burn up the VIP-155PT permanently.
- Either one power-source is allowed. Please make sure only one power source is applied to the VIP-155PT.

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Chapter 2

Preparations & Installation

Physical Installation Requirement

This chapter illustrates basic installation of VIP-155PT

- Network cables. Use standard 10/100BaseT network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

For Internet Access, an Internet Access account with an ISP, and either of a DSL or Cable modem (for WAN port usage)

Administration Interface

PLANET VIP-155PT provides GUI (Web based, Graphical User Interface) for machine management and administration.

Web configuration access:

To start VIP-155PT web configuration, you must have one of these web browsers installed on computer for management

- Netscape Communicator 4.03 or higher
- Microsoft Internet Explorer 4.01 or higher with Java support

Default LAN interface IP address of VIP-155PT is **192.168.0.1**. You may now open your web browser, and insert ***192.168.0.1*** in the address bar of your web browser to logon VIP-155PT web configuration page.

VIP-155PT will prompt for logon username/password, please enter: ***rootn*** / ***null (no password)*** to continue machine administration.

Note

Please locate your PC in the same network segment (192.168.0.x) of VIP-155PT. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

LAN/WAN Interface quick configurations

Nature of PLANET VIP-155PT is an IP Sharing (NAT) device, it comes with two default IP addresses, and default LAN side IP address is "**192.168.0.1**", default WAN side IP address is "**172.16.0.1**". You may use any PC to connect to the LAN port of VIP-155PT to start machine administration.

Hint

In general cases, the LAN IP address is the default gateway of LAN side workstations for Internet access, and the WAN IP of VIP-155PT are the IP address for remote calling party to connect with.

LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (default: **192.168.0.1**) of VIP in the adddress bar. After logging on machine with username/password (default: **root** / **null**), browse to "**Network**" --> "**LAN Config**" configuration menu:

Parameter Description

Bridge Mode	Enable this option to switch to bridge mode. VIP-155PT won't assign IP for its LAN port in bridge mode and its LAN and WAN port will be in the same network.
IP address	LAN IP address of VIP-155PT **Default: 192.168.0.1**
Subnet Mask	LAN IP address of VIP-155PT **Default: 255.255.255.0**
DHCP Service	Enable DHCP service in LAN port
NAT	Enable NAT function. If Bridge mode is enable, this function will be disabled.

ⓘ Hint

It is suggested to keep the DHCP server related parameters in default state to keep machine in best performance.

After confirming the modification you've done, Please click on the **Apply** button to macke the changes effective, browse to "**Config Manager**" --> "**Save Config**" configuration menu and click "**Save**" button to save configuration.

Then browse to "**System Manage**" --> "**Reboot**" configuration menu and click "**Reboot**" button to save configuration.

WAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (default: **172.16.0.1**) of VIP in the adddress bar. After logging on machine with username/password (default: **root** / **null**), browse to "**Network**" --> "**WAN Config**" configuration menu, you will see the configuration screen below:

Connection Type	**Data required.**
Static IP	The ISP will assign IP Address, and related information.
DHCP	Get WAN IP Address automatically; it is no need to configure the DHCP settings.
PPPoE	The ISP will assign PPPoE username / password for Internet access,

ⓘ Hint

Please consult your ISP personnel to obtain proper PPPoE/IP address related information, and input carefully.
If Internet connection cannot be established, please check the physical connection or contact the ISP service staff for support information.

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Chapter 3

Web Configurations

Configuring and monitoring your VIP-155PT from web browser

The VIP-155PT integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

Overview on the web interface of VIP-155PT

With web graphical user interface, you may have:

- More comprehensive setting feels than traditional command line interface.
- Provides user input data fields, check boxes, and for changing machine configuration settings
- Displays machine running configuration

To start VIP-155PT web configuration, you must have one of these web browsers installed on computer for management

- Netscape Communicator 4.03 or higher
- Microsoft Internet Explorer 4.01 or higher with Java support

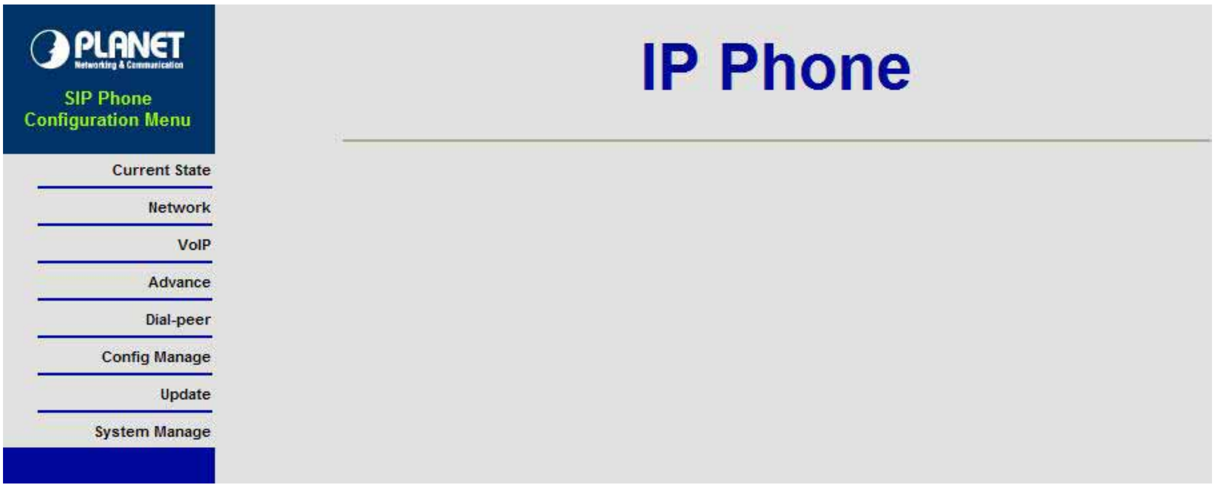
Manipulation of VIP-155PT via web browser

Log on VIP-155PT via web browser

After TCP/IP configurations on your PC, you may now open your web browser, and input ***http://192.168.0.1*** to logon VoIP gateway web configuration page.

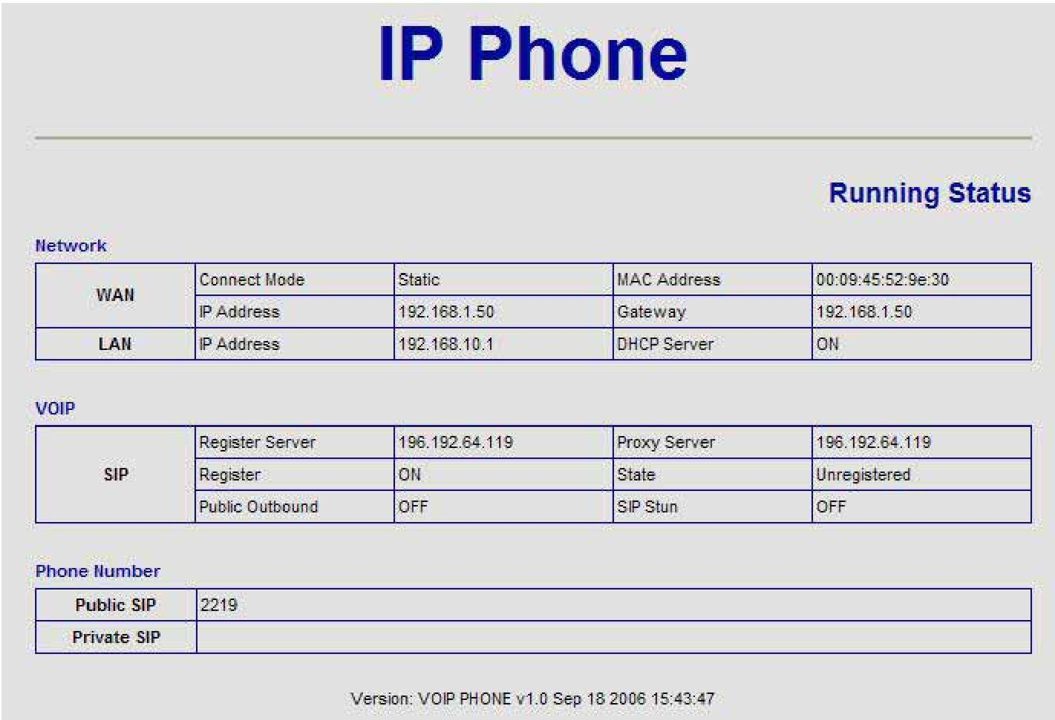
Browse any configuration menu, VIP-155PT will prompt for logon username/password, there are two level accounts for manage:

Account Name	Password	Level Description
root	**null** (no password)	Administrator user, can mansge all of configuration.
guest	**guest**	General user, just can manage part of configuration.



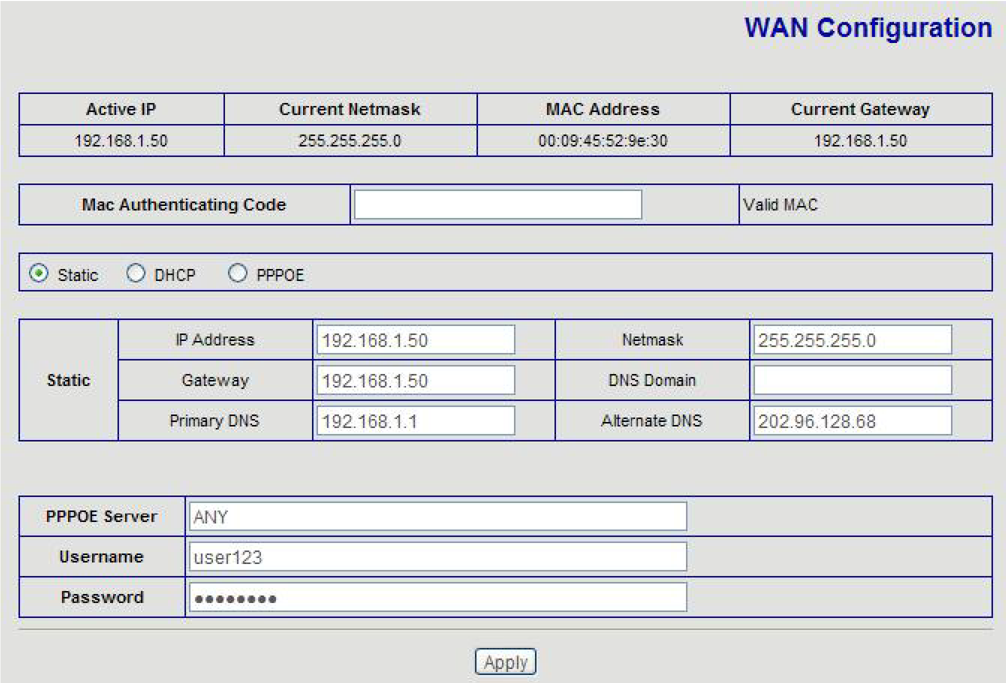
VIP-155PT main page

Current State



Current state information	
Network	Shows the WAN and LAN port connecting state and current settings
VOIP	Part show the working state of VoIP, you can see whether IP Phone has registered the public sip server
Phone Number	Shows the public sip and private sip phone numbers

WAN Config:



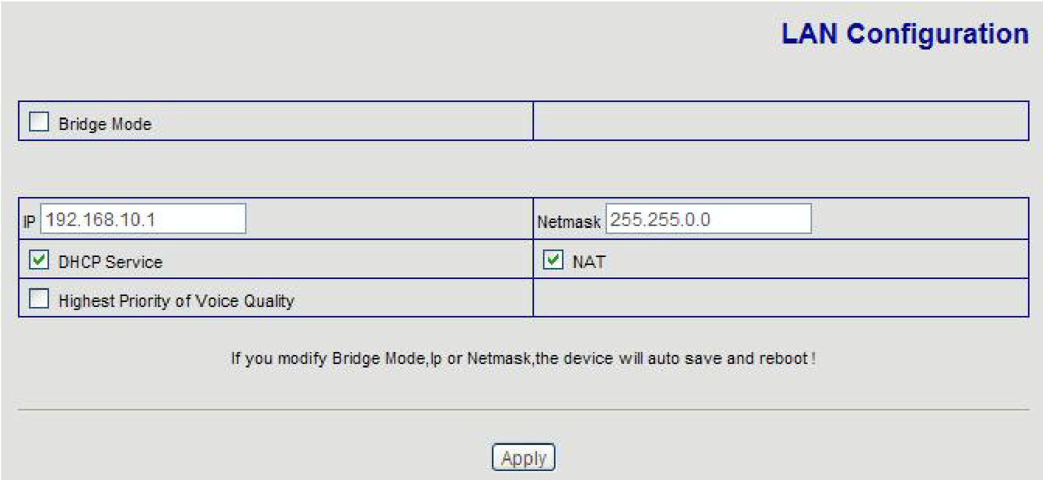
Three methods are available for Internet Access

Static IP	
Fixed IP User	If you are a leased line user with a fixed IP address, fill out the following items with the information provided by your ISP.
IP Address	check with your ISP provider
Netmask	check with your ISP provider
Default Gateway	check with your ISP provider

DHCP IP	
Dynmaic IP User	If there is DHCP server in your local network, VIP-155PT will automatically obtain WAN port network information from your DHCP server.

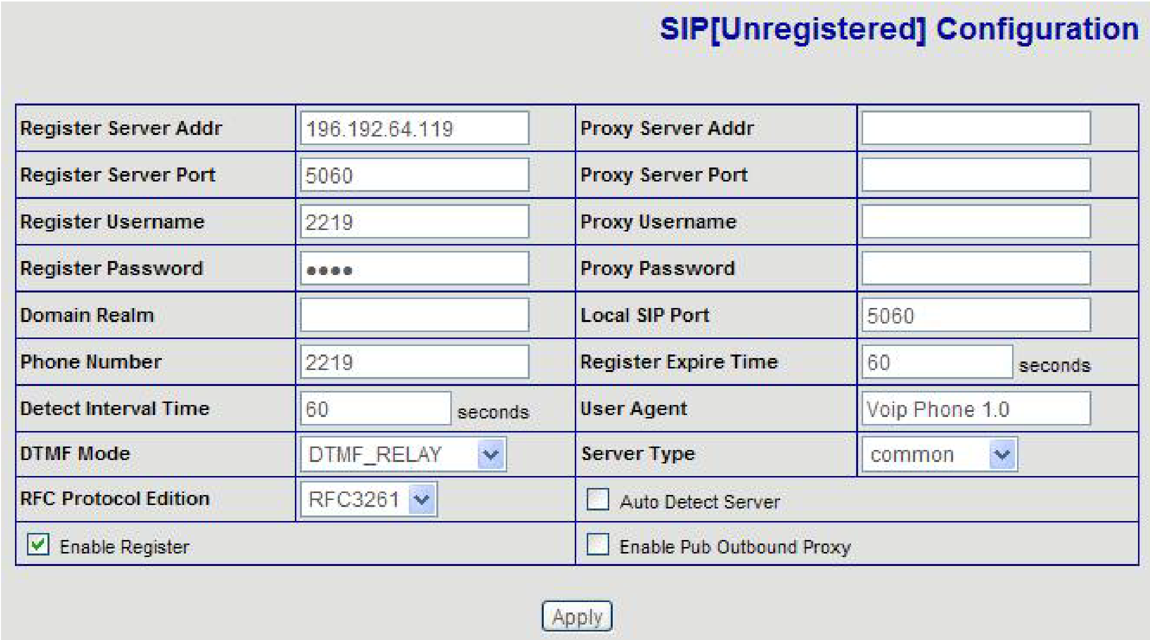
PPPoE	
PPPoE User	VIP-155PT will automatically obtain WAN port network information from your ITSP if PPPoE setting and the setup are correct.
PPPoE Server	Enter User Name provided by your ISP
Uasename	Enter Password provided by your ISP
Password	Enter Password to confirm again

LAN Config



Field	Description
Bridge Mode	Enable this option to switch to bridge mode. IP phone won't assign IP for its LAN port in bridge mode and its LAN and WAN port will be in the same network (This setting won't take effect unless you save the config and reboot the device)
IP address	LAN IP address of VIP-155PT **Default: 192.168.0.1**
Subnet Mask	LAN IP address of VIP-155PT **Default: 255.255.255.0**
DHCP Service	Enable DHCP service in LAN port
NAT	Enable NAT

SIP Config

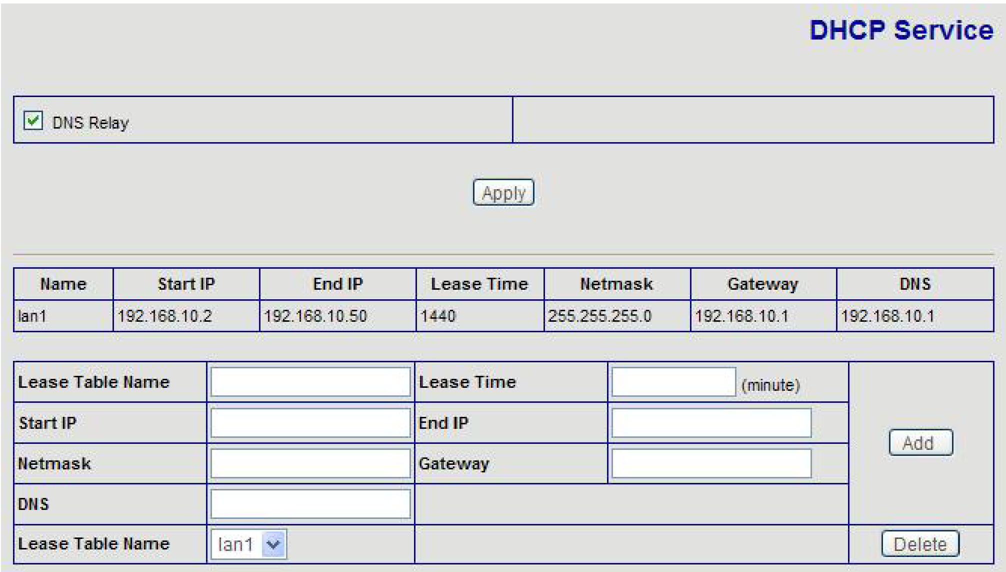


Setting page of public SIP server.

Field	Description
Register Server Addr	Register address of public SIP server
Register Server Port	Register port of public SIP server
Register Username	Username of your SIP account (Always the same as the phone number)
Register Password	Password of your SIP account
Proxy Server Addr	IP address of proxy SIP server (SIP provider always use the same IP for register server and proxy server, in this case you don't need to configure the proxy server information)
Proxy Server Port	Signal port of SIP proxy
Proxy Username	Proxy server username
Proxy Password	Proxy server password
Domain Realm	SIP domain, enter the sip domain if any, otherwise IP PHONE will use the proxy server address as sip domain
Local SIP port	Local SIP register port, default 5060
Phone Number	Phone number of your SIP account
Register Expire Time	Register expire time, default is 600 seconds. IP PHONE will auto configure this expire time to the server recommended setting if it is different from the SIP server
Detect Interval Time	Co-work with the *Auto Detect Server*, if *Auto Detect Server* is enable, IP PHONE will periodically detect if the SIP server is available according this setting
User Agent	It will show IP Phone's information on Proxy Server

DTMF Mode	DTMF signal sending mode: support RFC2833, DTMF_RELAY (inband audio) and SIP info
Server Type	It could support different SIP Proxy providers
RFC Protocol Edition	Current IP PHONE SIP version. Set to RFC 2543 if the gate need to communicate to devices (such as CISCO5300) using the SIP 1.0. Default is RFC 3261
Auto Detect server	Co-work with *Server Auto Swap* and *Detect Interval Time*. Enable this option, IP PHONE will periodically detect whether the public SIP server is available, if the server is unavailable, the IP PHONE will switch to the back-up SIP sever, and continue detecting the public sip server. IP PHONE will switch back to the primary SIP server if the server is available again
Enable Register	Enable/Disable SIP register. IP PHONE won't sent register info to SIP server if disable register
Enable Pub Outbound Proxy	Enable/Disable Outbound Proxy

DHCP Server



DHCP server manage page.

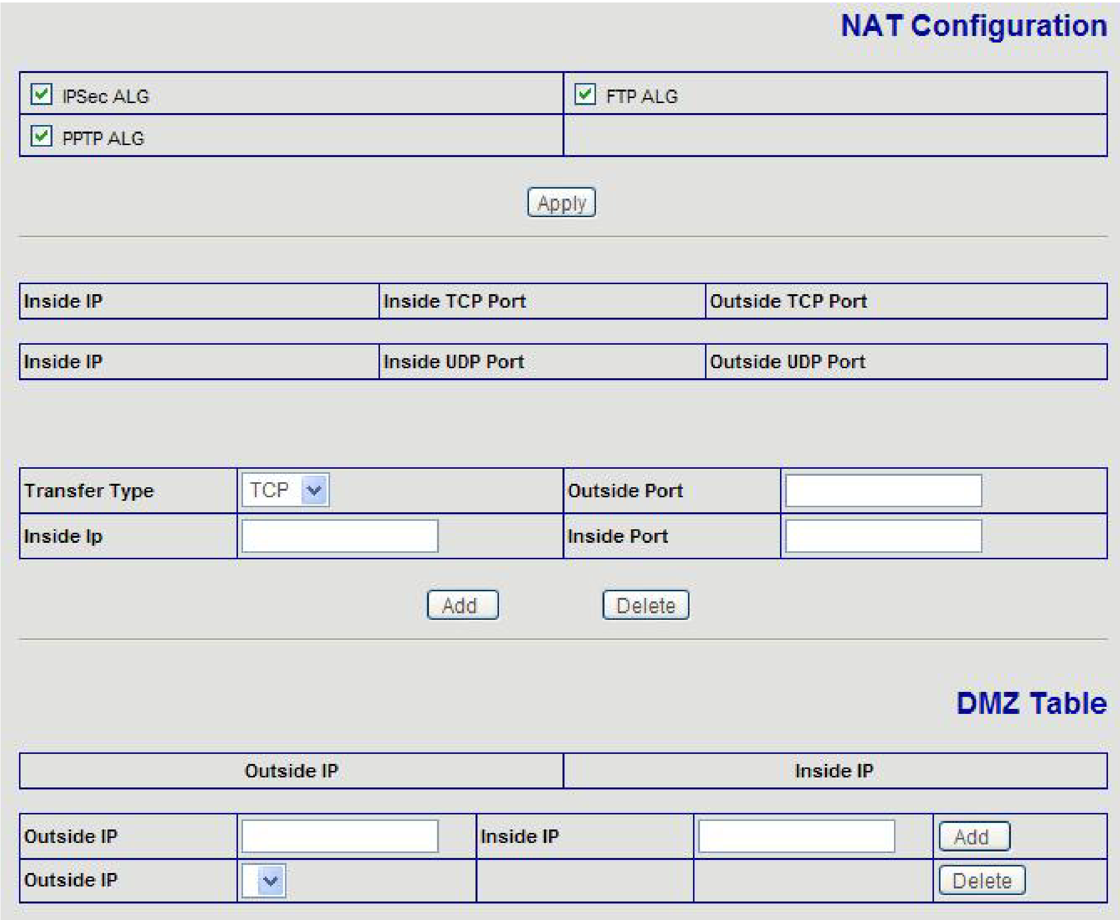
User may trace and modify DHCP server information in this page

Field	Description
DNS Relay	Enable DNS relay function
Lease Table Name	Lease table name
Lease Time	DHCP server lease time

Start IP	Start IP of lease table
End IP	End IP of lease table. Network device connecting to the IP PHONE LAN port can dynamic obtain the IP in the range between start IP and end IP
Netmask	Netmask of lease table
Gateway	Default gateway of lease table
DNS	Default DNS server of lease table

Notice: This setting won't take effect unless you save the config and reboot the device

NAT



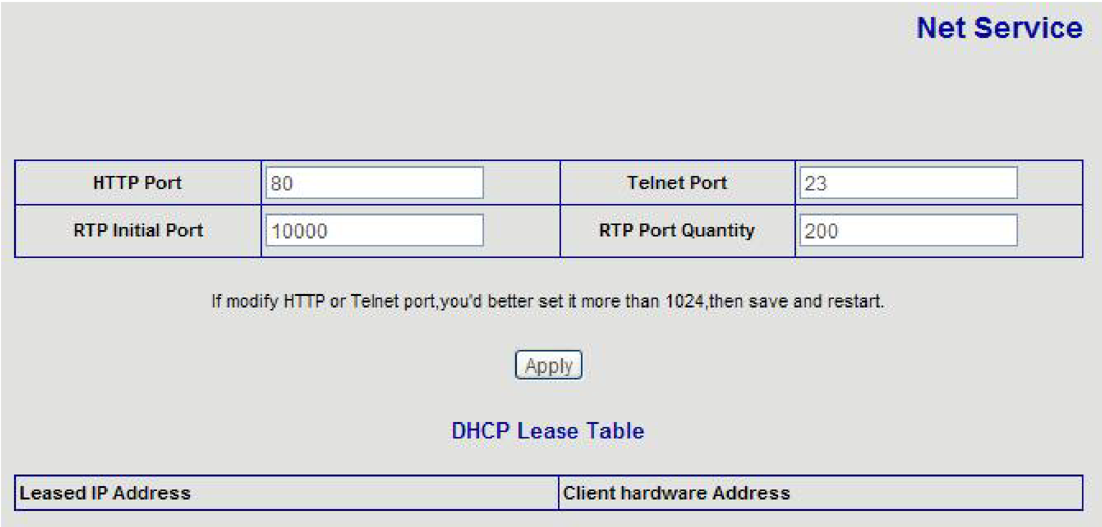
Advance NAT setting. Maximum 10 items for TCP and UDP port mapping.

Field	Description
IPSec ALG	Enable/Disable IPSec ALG
FTP ALG	Enable/Disable FTP ALG
PPTP ALG	Enable/Disable PPTP ALG
Transfer Type	Transfer type using port mappin

Inside IP	LAN device IP for port mapping
Inside Port	LAN device port for port mapping
Outside Port	WAN port for port mapping

Click Add to add new port mapping item and Delete to delete current port mapping item.

Net Service



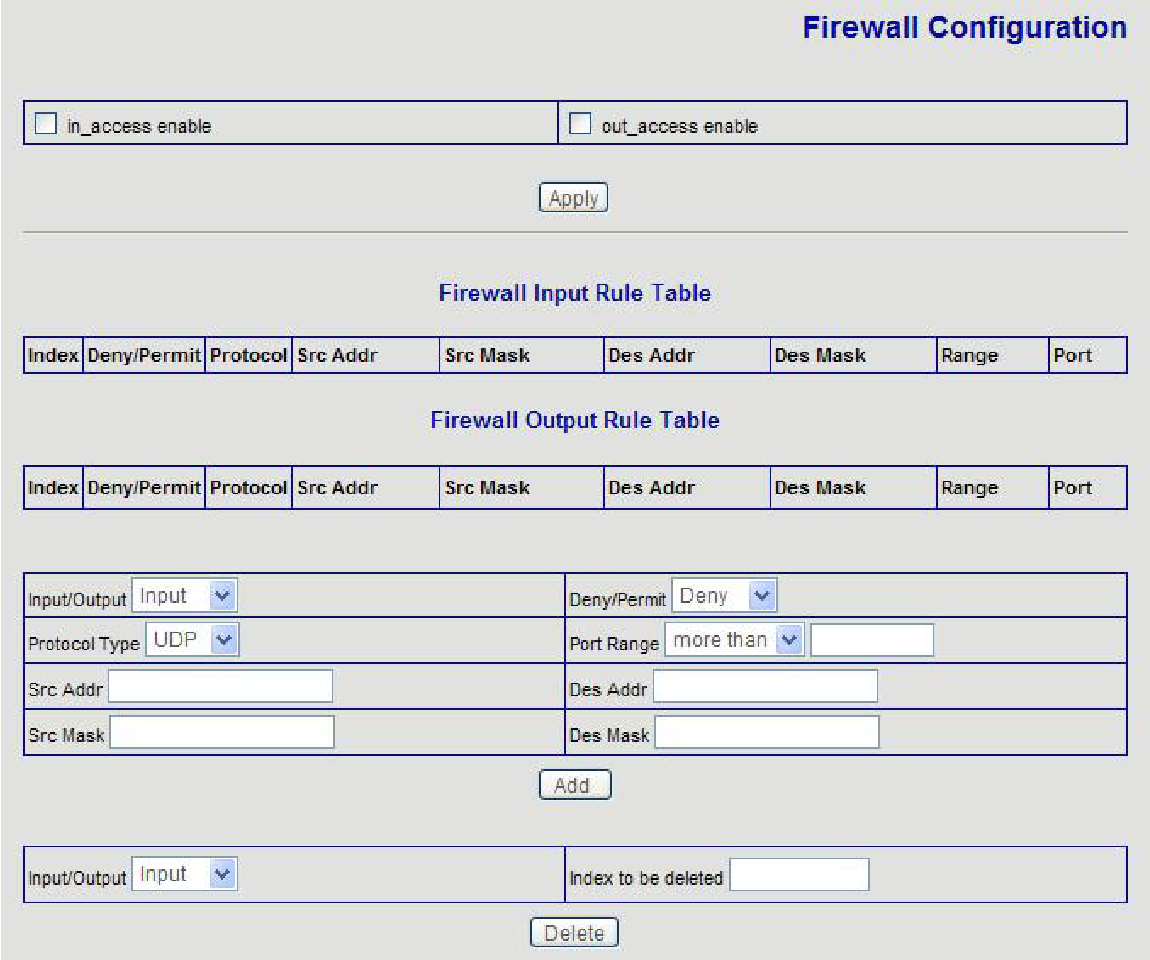
Field	Description
HTTP Port	Configure HTTP transfer port, default is 80.User may change this port to enhance system's security. When this port is changed, please use http://xxx.xxx.xxx.xxx:xxxx/ to reconnect.
Telnet Port	Configure telnet transfer port, default is 23
RTP Initial Port	RTP initial port
RTP Port Quantity	Maximum RTP port quantity, default is 200

Notice: Settings in this page won't take effect unless save and reboot the device.

If you need to change telnet port or HTTP port, please use the port greater than 1024, because ports under 1024 is system remain ports.

HTTP service if HTTP is set to 0.

Firewall settings



Firewall setting page. User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the internet.

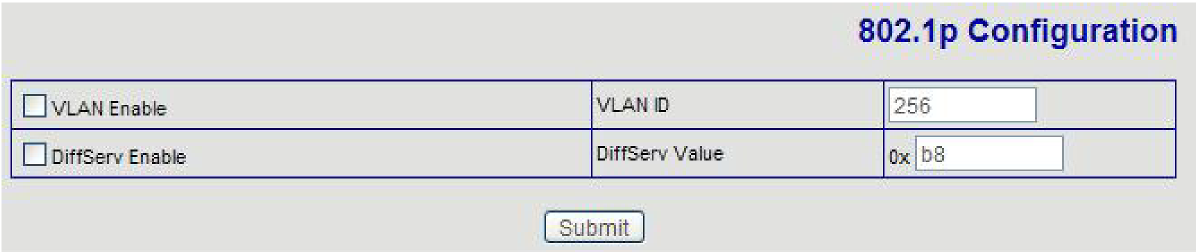
Access list support two type limits: input_access limit or output_access limit. Each type support 10 items maximum.

IP PHONE firewall filter is base WAN port. So the source address or input destination address should be WAN port IP address.

Field	Description
in_access enable	Enable in_access rule
out_access enable	Enable out_access rule
Input/Output	Specify current adding rule is input rule or output rule
Deny/Permit	Specify current adding rule is deny rule or permit rule
Protocol Type	Protocol using in this rule: TCP/IP/ICMP/UDP
Port Range	Port range if this rule
Src Addr	source address. Can be single IP address or network address

Dest Addr	destination address. Can be IP address or network address
Src Mask	source address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID
Des Mask	Destination address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID

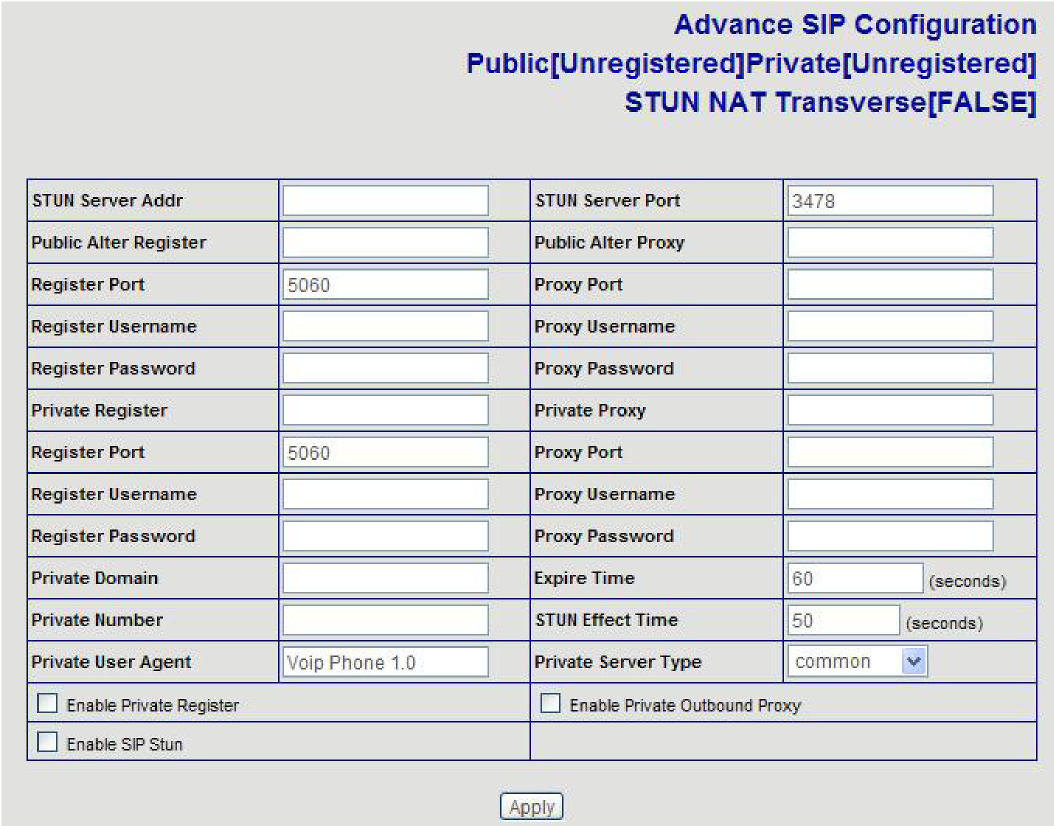
QoS settings



IP PHONE IP phone implement QoS based on 802.1p, The QoS is used to mark the network communication priority in the data link/MAC sub-layer. IP Phone will sorted the packets using the QoS and sends it to the destination.

Field	Description
VLAN Enable	If enable the VLAN service, the second layer will realize separate voice, signal and data transmission. To realize separate voice and data transmission by dispose for IP precedence of ToS area of voice transmission. To reach upper layer switch or router have priority to transfer voice transmission. (The prerequisite is the upper layer swirch or router have to identify ToS area.)
VLAN ID	Dispose VLAN ID is add a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 1~4094.
DiffServ Enable	If enable the VLAN service, it indicates use DSCP mode to realize three layers QoS. This moment, the DSCP of SIP signals which between IP Phone and MGC. It will use Class Selector 5 (The value is 0xA0). And the DSCP of mediums information (In RTP packets) would be used the values of DiffServ Value field.
DiffServ Value	The value range are 00 ~ FF. (0x28, 0x30, 0x38, 0x48, 0x50, 0x58, 0x68, 0x70, 0x78, 0x88, 0x90, 0x98, 0xb8)

Advance SIP settings

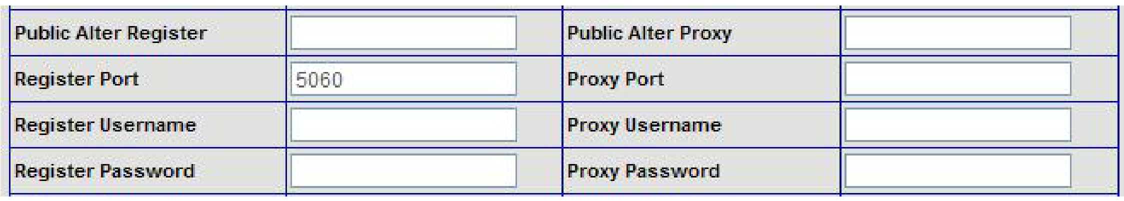


This page is used to set the private sip server, stun server, and back up sip server information.

STUN Server setting:

Field	**Description**
STUN Server Addr	Configure stun server address
STUN Server Port	Configure stun server port default 3478
STUN Effect Time	Stun detect NAT type circle, unit: minute
Enable SIP STUN	Enable/disable stun

Public Alter Register:



Public Alter server provide redundancy for the public server, if the public server is unavailable, IP PHONE will use the alter server, and switch back to the public server when it is available. Account setting in public alter setting should be the same as the public server.

Please refer to SIP_Config for the setting for how to set the public alter server.

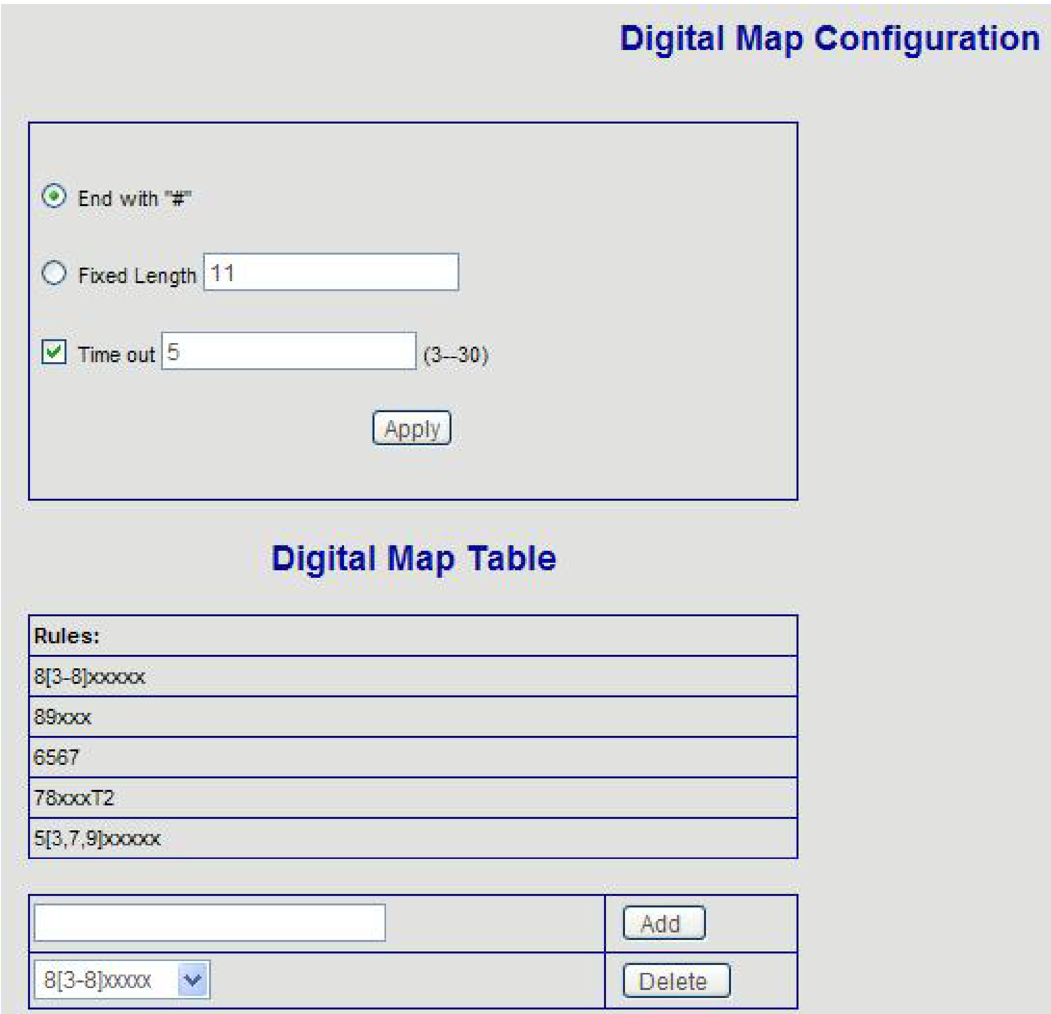
User can register two sip servers:

Private Register		Private Proxy	
Register Port	5060	Proxy Port	
Register Username		Proxy Username	
Register Password		Proxy Password	
Private Domain		Expire Time	60 (seconds)
Private Number		STUN Effect Time	50 (seconds)

Public sip server and private sip server.these two sip servers are independent from each other and running in the same time.

For how to configure private sip server. Please refer to SIP_Config

Digital Map



Digit map is a set of rules to determine when the user has finished dialing.

IP Phone support below digital map:

Digital Map is based on some rules to judge when user end their dialing and send the number to the server.

VIP-155PT support following digital map:

Field	Description
End With "#"	Use # as the end of dialing
Fixed Length	When the length of the dialing match, the call will be sent
Timeout	Specify the timeout of the last dial digit. The call will be sent after timeout

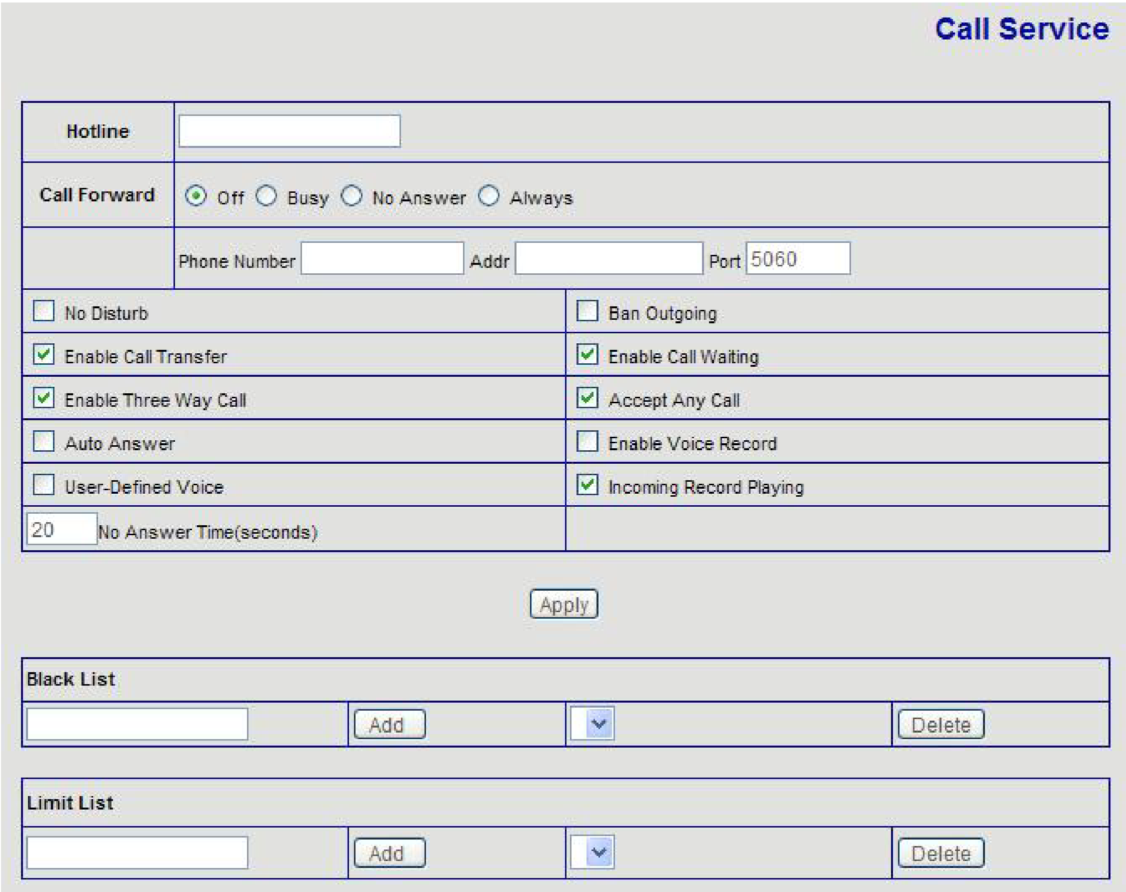
User Define digital map:

Field	Description
[ ]	Represents the range of digit, can be a range such as [1-4], or use comma such as [1,3,5], or use a list such as [234]
x	Represents any one digit between 0~9
Tn	Represents the last digit timeout. n represents the time from 0~9 second, it is necessary. Tn must be the last two digit in the entry. If Tn is not included in the entry, we use T0 as default, it means system will sent the number immediately if the number matches the entry.

Example:

Field	Description
[1-8]xxx	All number from 1000 to 89999 will be sent immediately
9xxxxxxx	8 digits numbers begin with 9 will be sent immediately
911	Number 911 will be sent will be immediately
99xT4	3 digits numbers begin with 99 with be sent after four seconds

Call Service Settings



User configure the value add service such as hotline, call forward, call transfer, 3-way conference call .etc in this page

Field	Description
Hotline	Configure hotline number. IP PHONE immediately dials this number after hook-off if it is set
Call Forward	**Forward when busy:** select *Busy* in the *Call Forward* Field, and Key in the destination phone number in the *Forward Number*. If some one calls you when you having a call, the caller will be forwarded to the destination number. **Forward no answer:** Select *No Answer* in the *Call Forward* Field, and Key in the destination phone number in the *Forward Number*, fill the time in the *No Answer Time.* If some one calls you and no one answer the caller during the No Answer Time, the call will be forward to the destination number. **Forward Always:** Select *Always* in the *Call Forward* Field, and Key in the destination phone number in the *Forward Number*, then any one call this gateway will be forward to the destination number.
No Disturb	DND, do not disturb, enable this option to refuse any calls

Ban Outgoing	Enable this to ban outgoing calls
Enable Call Transfer	Check the *Enable Call Transfer*. **Unattended transfer:** If A is the IP PHONE user, and B calls and talking with A through VoIP. A can **press FWD button** to hold the call with B, and then **enter C's number**. B will be transferred to C and can talk with C. **Attended transfer:** If A is the IP PHONE user, and B calls and talking with A through VoIP. A can **press Hold button** to hold the call with B, and then **enter C's number** to talk will C. and press **Hold** to switch back to A, and then press **FWD** key , B will be transferred to C and can talk with C.
Enable Call Waiting	Enable/disable Call Waiting
Enable Three Way Call	Check Enable Three Way Call Assume A is the VIP-155PT user, and B calls and talking with A through VoIP. A can **press FWD button** to hold the call with B, then **enter *** and then **enter C's number** to talk with C, and then **press * button** again to make 3-way conference calls.
Accept Any Call	If this option is disable, IP PHONE refuse the incoming call when the called number is different from IP PHONE's phone number.
Auto Answer	Enable/disable auto answer function
Enable Voice Record	Enable/disable answering machine function. Please refer to the bwloe descriptions for detail.
User-defined Voice	Use customized greeting message
Incoming Record Playing	Simultaneously play the message when recording
No Answer Time	No answer call forward time setting
Black List	Incoming call in these phone numbers will be refused
Limit List	Outgoing calls with these phone numbers will be refused

Voice Record

VIP-155PT provides record function. With this function, user may record three VoIP message and one local message.

Field	**Description**
Enable Voice Record	Select **"Enable Voice Record"** to active answering machine, and config **No Answer Time.** If there is an incoming call and no one answer the call. After timeout, IP PHONE will auto answer this call and ask the caller to leave message.

Incoming Record Playing	Play the message when recording
User-Defined Voice	Use customizes greeting voice for answering machine

Record local message:

User may use local message to leave message to other local users.

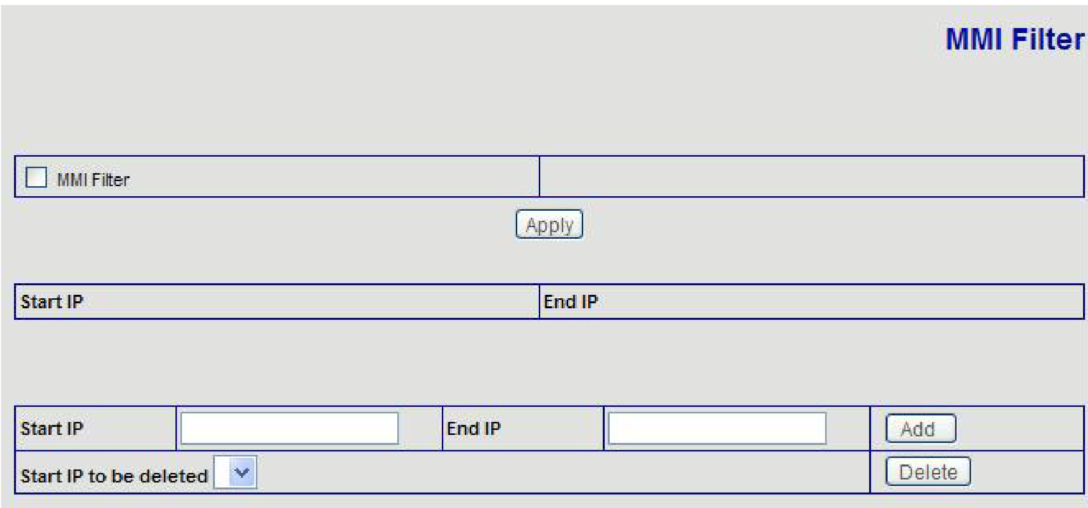
Please refer the **Record** button function as below:

Record Function		
Level1	Level2	Description
Received	New	New message info
	Old	Old message info
	Record	Enable/disable answering machine
	Playing	Enable/disable Incoming Record Playing
Local	Play	Play local message
	Rec	Record local message
User define	Switch	Enable/disable customize greeting message
	Play	Play customize greeting message
	Rec	Record customize greeting message

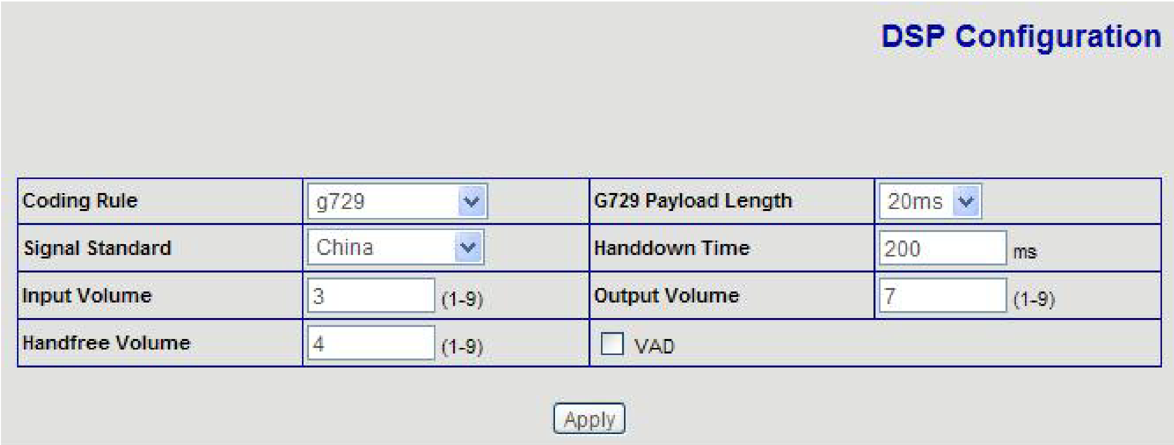
MMI Filter

MMI filter is used to make access limit to IP PHONE IP phone.

When MMI filter is enable. Only IP address within the *start IP* and *end IP* can access IP PHONE IP phone.



Audio Settings



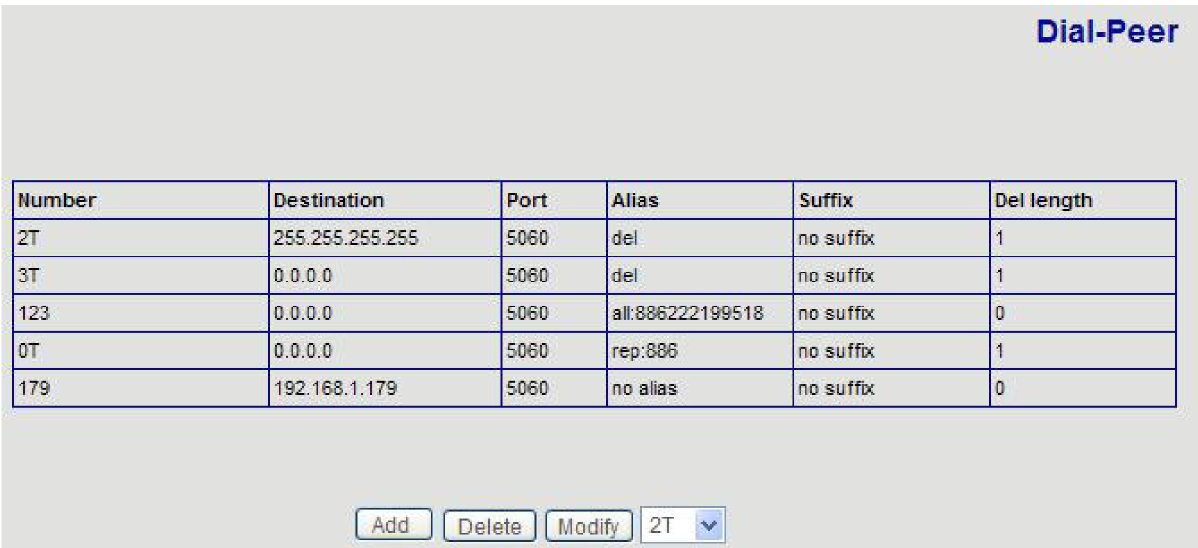
DSP Configuration

Coding Rule	g729	G729 Payload Length	20ms
Signal Standard	China	Handdown Time	200 ms
Input Volume	3 (1-9)	Output Volume	7 (1-9)
Handfree Volume	4 (1-9)	VAD	

Apply

Field	Description
CODEC	select the prefer CODEC; support ulaw, alaw,G729 and G7231 5.3/6.3
Signal Standard	Signal standard for different area
Input Volume	Handset in volume
Output Volume	Handset out volume
Handfree Volume	Hand free volume
Handdown Time	hand down detect time
G729 Payload Length	G729 payload length
VAD	Enable/disable Voice Activity Detection

Dial-Peer Settings



Dial-Peer

Number	Destination	Port	Alias	Suffix	Del length
2T	255.255.255.255	5060	del	no suffix	1
3T	0.0.0.0	5060	del	no suffix	1
123	0.0.0.0	5060	all:886222199518	no suffix	0
0T	0.0.0.0	5060	rep:886	no suffix	1
179	192.168.1.179	5060	no alias	no suffix	0

Add Delete Modify 2T

VIP-155PT provide flexible dial rule, with different dial-rule configure, user can easily implement the following function:

----Replace, delete or add prefix of the dial number.

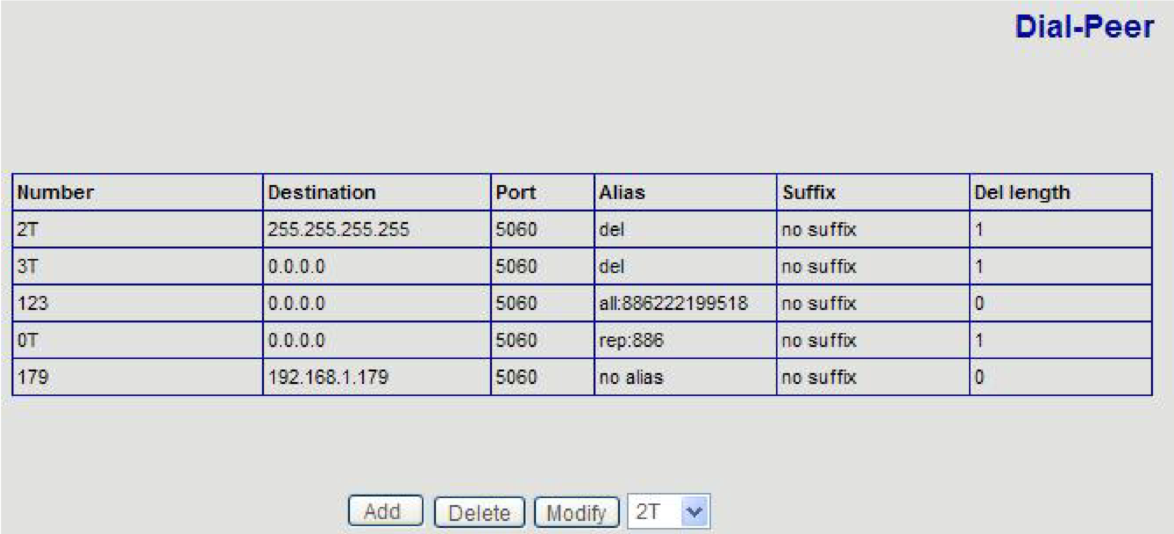
----Make direct IP to IP call

----Place the call to different servers according the prefix.

You can click "Add" to add a new dial rule. Below is the detail setting of the dial-rule:

Field	Description
Phone Number	The Number suit for this dial rule, can be set as full match or prefix match. Full match means that if the number user dialed is completely the same as this number, the call will use this dial-rule. Prefix match means that if prefix of the number that the user dials is the same as the prefix, the call will use this dial-rule, to distinguish from the full match case, you need to add "T" after the prefix number in the phone number setting
Destination (optional)	call destination, can be IP or domain. Default is 0.0.0.0, in this case the call will be routed to the Public SIP server. If you set the destination to 255.255.255.255, then the call will be routed to the private SIP server. Also you can key other address here to make direct IP calls
Port (optional)	Configure the port of the destination, default is 5060 in SIP and 1720 for H323
Alias (optional)	Set up the Alias. We support four Alias as below. Alias need to co-work with the *Del Length*: ➢ add:xxx, add prefix to the phone number, can set to reduce the dial length. ➢ all: xxx, replace the phone number with the xxx, can use as speed dial function. ➢ del, delete the first N numbers. N is set in the *Del Length* rep:xxx, replace the first N numbers. N is set in the Del Length. For Example: Use wants to place a call 8610-62281493, then you can set the *phone number* in the dial rule as 010T, and set the *Alias* as rep:8610, and set the *Del Length* to 3. Then all calls begin with 010 will be changed to 8610 xxxxxxxx.
Suffix (optional)	Configure suffix, show no suffix if not set

Example:



Number	Destination	Port	Alias	Suffix	Del length
2T	255.255.255.255	5060	del	no suffix	1
3T	0.0.0.0	5060	del	no suffix	1
123	0.0.0.0	5060	all:886222199518	no suffix	0
0T	0.0.0.0	5060	rep:886	no suffix	1
179	192.168.1.179	5060	no alias	no suffix	0

Field	Description
2T rule	If the call starts with 2, the first 2 will be deleted, and the rest number will be sent to private SIP server.
3T rule	If the call starts with 3, the first 3 will be deleted, and the rest number with be sent to public SIP server.
123 rule	Dial 123 and will send 8675583018049 to your server. Used as speed dial function
0T rule	If the calls is begin with 0, the first 0 will be replace by 86. Means that if you dial 075583018049 and AG-188 will send 8675583018049 to your server.
179 rule	When you dial 179 , the call with send to 192.168.1.179, suit for LAN application without set up a sip server.

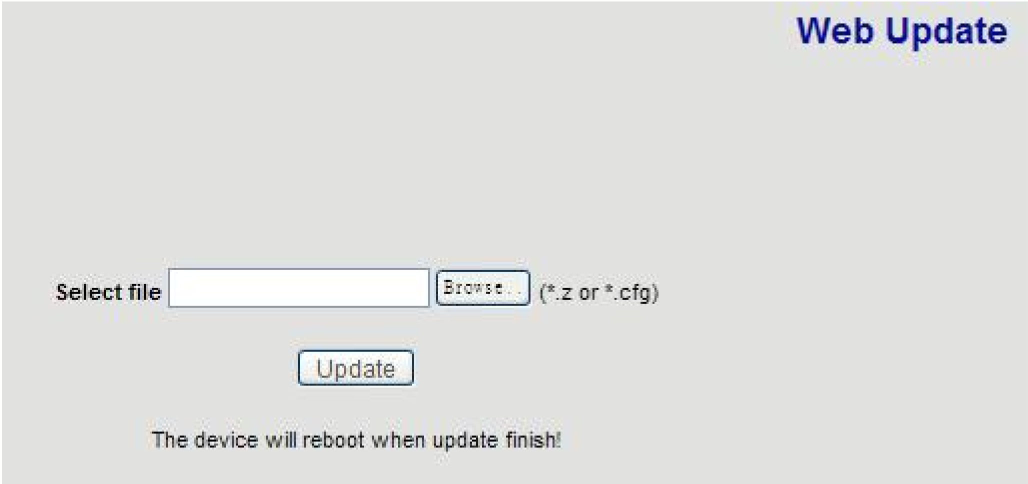
Config Manage

Field	Description
Save Config	Save current settings
Clear Config	Restore to default settings

Notice: clear config in admin mode, all settings restores to factory default; clear config in guest modem, all settings except sip and advance sip restore to factory default.

WEB Update

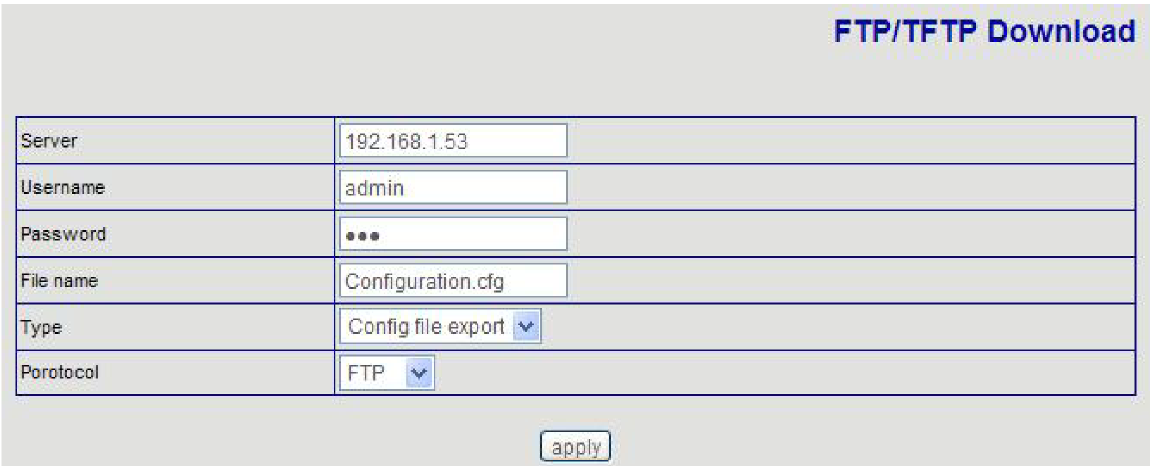
Update IP phone's settings or firmware. Firmware file is .z extension when configure file is .cfg extension, IP PHONE will auto select configure update or firmware update according the extension.



FTP/TFTP Update

Backup:

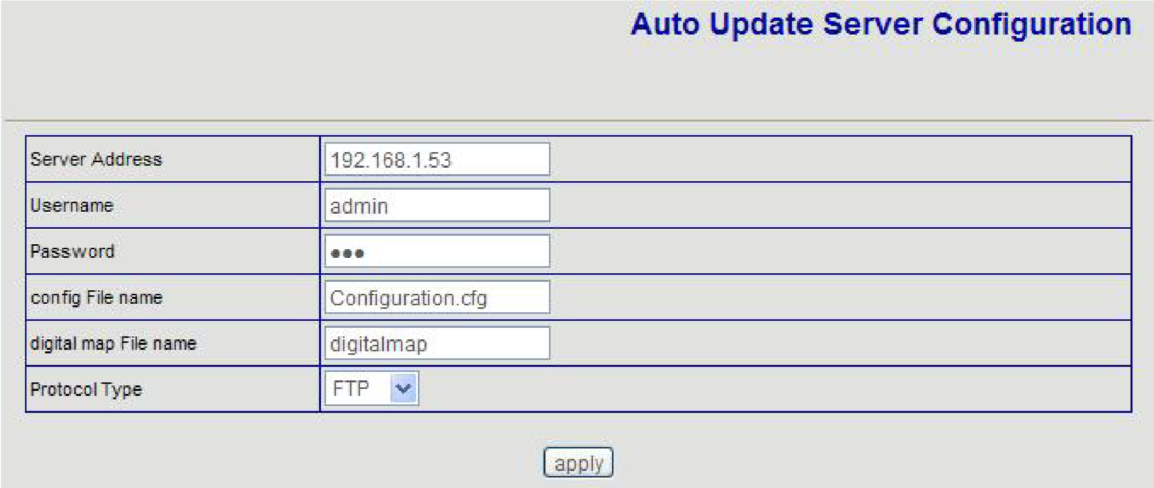
Back up configure file to your FTP/TFTP server.



* configure use .cfg extension.

Auto update:

IP PHONE IP phone support FTP and TFTP auto update. The gateway will auto obtain the configure file from your update server if configured. To obtain the original configure file, you can use the FTP/TFTP back up as describe above. Configure file using module structure, user may remain the concerned modules and remove other modules. Put the configure file in the root directory of update serve when finish editing.



Configure file version was in the <<VOIP CONFIG FILE>> and <GLOBLE CONFIG MODULE> ConfFile Version

Example:

Gateway original version is:

<<VOIP CONFIG FILE>>Version:1.0000

<GLOBLE CONFIG MODULE>　ConfFile Version：6

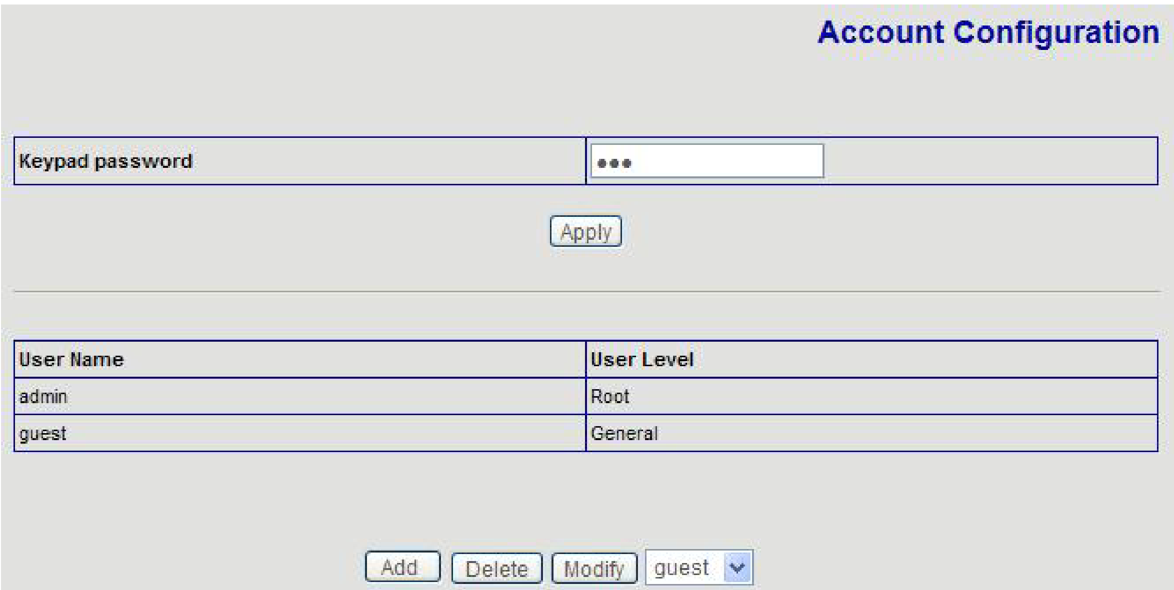
User may edit the configure file version to:

<<VOIP CONFIG FILE>>Version:1.0007

<GLOBLE CONFIG MODULE>　ConfFile Version：7

Account Manage

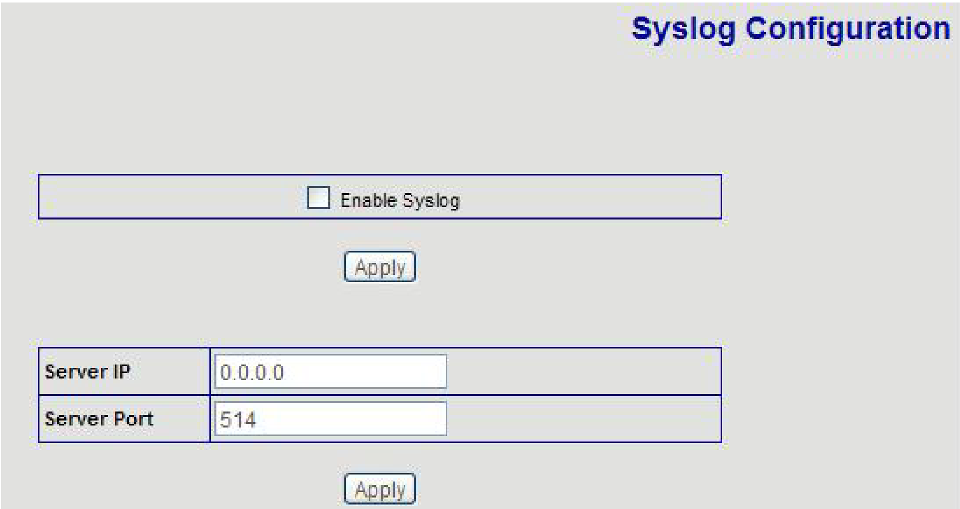
Set web access account or keypad password of IP PHONE.



Phone Book

User may set contacts in this page, and the contacts will be saved in the memory. Then using the Pbook, Vol+,Vol-,Menu/OK and Exit keys to choose your friend in the contacts and then press # to call out.

Syslog Config



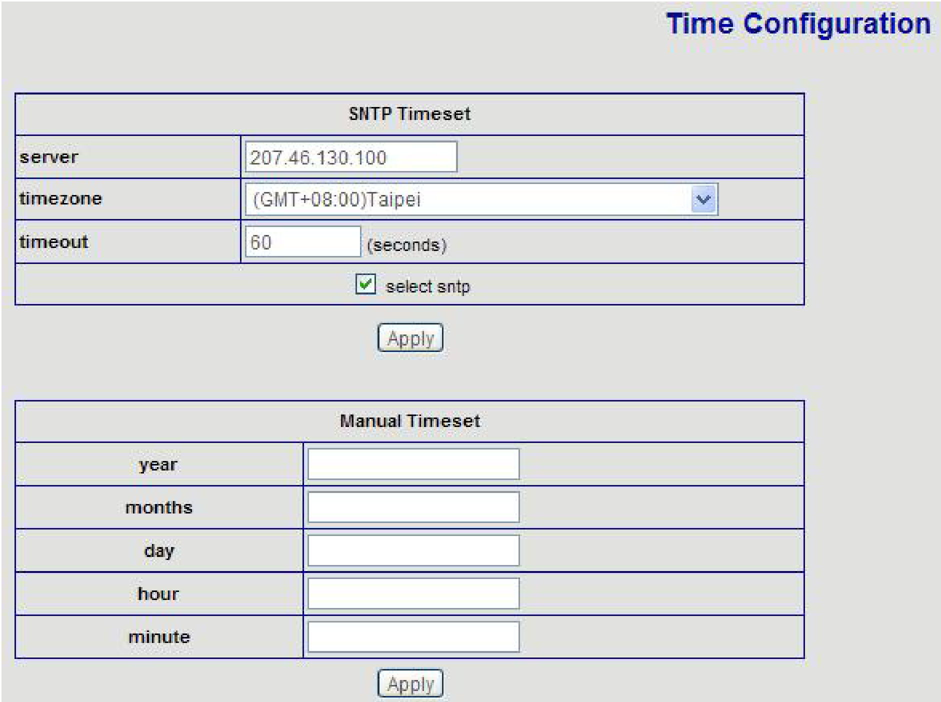
Field	Description
Enable Syslog	Enable syslog function.
Server IP	VIP-155PT will automatic send the system logs to define server. Fill in the server IP address.
Server Port	Fill in the transmission port.

Time Set

VIP-155PT could support SNTP timeset, type in SNTP Server address, Timezone and timeout fileds.

And click “**select sntp**” to enable SNTP function.

If hasn’t click “**select sntp**”, it also could set up time by manual.



Reboot

Reboot IP phone, some setting needs to reboot to make it works. Please always save config before reboot, otherwise the setting will return to previous setting.

4

Chapter 4

Keypad Configurations

Keypad Function

User can configure IP PHONE through its keypad. List below is the keypad function

Keypad	**Mode**	**Function/Display**
Idle mode	----	show current time
Sysinfo	Idle mode	circularly show phone number,wan ip, gateway info
Menu/OK	Idle mode	enter config mode, default password 123
	config mode	confirm or enter sub-menu
Exit	config mode	exit
Up	Calling mode	volume up (Max:9)
	config mode	Page up
Down	Calling mode	volume down (Min:1)
	config mode	Page down
Del	Calling mode	Delete digits
	config mode	Delete digits
Mute	Calling mode	Mute
Out call	Idle mode	Outgoing call menu
In call	Idle mode	Incoming call menu
Record	Idle mode	Enter record menu, usage refer FAQ
Pbook	Idle mode	Enter Phone book set up
Handfree	Calling mode	Handfree
0 - 9	Calling mode	Digits 0~9
	config mode	Hit quickly to switch between numeric or alphabetic
*	Calling mode	Use in 3-way conference call.
	config mode	Use as "." In the ip address setting

#	Calling mode	Use as end key of dialing or the dial number
Hold	Calling mode	Hold, detail refer value add service
FWD	Calling mode	Transfer, detail refer value add service
Redial	Calling mode	Redial key
Send	Calling mode	call key
No.1~No.9	Idle mode	Speed dial key

Keypad Menu

User may use **SET, Menu/ok, Exit, Vol+**, **Vol-** to config IP PHONE detail setting. Press **Menu/ok** to enter config mode, and the default password is **123**.

Below list the keypad menu of IP PHONE

IP PHONE Keypad Menu			
Level 1	Level 2	Level 3	Level 4
Network	LAN	IP	
		Netmask	
		DHCP Server	
		NAT	Switch
			FTPalg
			IPSec alg
			PPTPalg
		Bridge Mode	
	WAN	Status	
		Static Net	1. IP
			2. NetMask
			3. Gateway
			4. DNS
			5. DNS2
		PPPoE	User name
			Password
		QoS	
Call Feature	Phone-number	Public SIP	
		Private SIP	
	Limit-List	Current	
		ADD	
		DEL	
	Black-List	Current	
		ADD	

		DEL	
	FastCall		
	Three Call		
	Call-Transfer		
	Call-Waiting		
	Call-Forward	Condition	
		SIP	Transfer Num
			Transfer IP
			Port
	Dial-Rule	End With #	
		Fixed Length	Switch
			Length
SIP	Reg Status	Public Reg	
		Private Reg	
	Reg Switch	Public	
		Private	
	Server	Private	Register
			Proxy
	Domain	Public	
		Private	
	User Agent	Public	
		Private	
	Detect-server		
	Dtmf-mode		
	Interval-time		
	Swap-server		
	RFC-version		
	Signal-Port		
	Stun	Switch	
		Addr	
		Port	
		Effect Time	
DSP	Codec		
	Handdown-time		
	Dtmf-Volume		
	Input-volume		
	Output-Volume		
System	Save		
	Reboot		

	Set Default	
Other Setting	Syslog	Switch
		Server-IP
		Server-Port

5

Chapter 5

Telnet Console

Introduce

Basic Structure

User may use telnet command to access and manage IP phone.

IP PHONE adopts tree structure for telnet. Every node contains its sub-nodes or local command. User can type "help" or "?" whenever to see sub-nodes and all local command under current node.

Besides local command, there are some global commands can be used in each node.

Basic command

Logout: exit telnet mode.

Write: save current settings.

Type sub-nodes name in current node to switch to sub-node.
Type "!" or "exit" in current node to return to parent-node.

Type "help" or "?" can see all sub-nodes and all local command under current node, every help item has comments such as <command> or <node> to distinguish sub-nodes and local command. Type "help" or "?" in command can see all parameters using in this command.

When typing node name or command, user no need to key the full name, use TAB button will make it more efficient.

There are two types in command parameters: optional and required. "required" parameter use "-" as prefix and "optional" use "_" as prefix. User may type "-" or "_" then press TAB button for complementarily.

Global Command

Global command is available under all nodes, IP PHONE support following commands:

Command	Function	Example
chinese	Set to Chinese UI	#chinese
clear	Clear telnet screen	#clear
english	Set to English UI	#english
exit	Return to parent-node	#exit
help	1. Show help info 2. Show sub-nodes and local command	1. #help ping 2. #help
history	Show command history	#history
logout	Exit	#logout
ping	Ping command, use to check network,	#ping www.google.com
tree	Print tree structure of current command	#tree
who	Show current user	#who
write	Save setting to flash	#write

Tree Structure

account

path: <account>#

[stop]start Syslog ---syslog [no] start

Configure Syslog server address and port ---syslog server –ip x.x.x.x _port xxx

Example: #<config-account-syslog>#server ---ip 202.112.20.10

Show syslog settings ---syslog show

Show all account settings ---show

config

➢ **accesslist firewall config**

path: <config-accesslist>#

add firewall rule ---entry –I/O xxx –P/D xxx –proto xxx –srcaddr x.x.x.x –srcmask x.x.x.x–desaddr x.x.x.x –desmask x.x.x.x –portrange xxx –portnum xxx

Example:<config-accesslist>#entry –I/O input –P/D deny –proto udp –straddr 202.112.10.1 –srcmask 255.255.255.0 –desaddr 210.25.132.1 –desmask 255.255.255.0 –portrange neq –portnum 5060 delete firewall rule ---no entry –I/O xxx –index xxx

Example :<config-accesslist>#no entry –I/O input –index 1

Show firewall settings ---show

[disable] enable input filter ---[no]in-access

[disable] enable output filter ---[no]out-access

- **DHCP**

path: <config-dhcp>#

add DHCP rule ---entry –name xxx –startip x.x.x.x –endip x.x.x.x –netmask x.x.x.x –gateway x.x.x.x –dnsserver x.x.x.x _time xxx

Example:<config-dhcp>#entry –name lan2004 –startip 192.168.1.2 –endip 192.168.1.254 –netmask 255.255.255.0 –gateway 192.168.1.1 –dnsserver 192.168.10.18

delete DHCP rule ---no entry –name xxx

Example: <config-dhcp>#no entry –name lan2004

Show DHCP settings ---show

[disable]enable DNS-relay ---[no]dns-relay

- **dialrule**

path: <config-dialrule>#

[disable] enable End with # ---[no]endchar

Set end with fix length ---fixlen xxx

Disable end with fix length ---no fixlen

Set timeout to send ---timeout-send xxx

Disable timeout to send ---no timeout-send

Add digital map ---entry –prefix xxx –length xxx

Example: <config-dialrule>#entry –prefix 010 –length 11

Delete digital map rule ---no entry –prefix xxx

Example: <config-dialrule>#no entry –prefix 010

Show current digital map ---show

- **LAN interface settings**

path: <config-interface-fastethernet-lan>#

[disable]enable bridge mode ---[no]bridgemode

[disable]enable DHCP service ---[no]dhcp-server

[disable]enable NAT ---[no]nat

Show current DHCP rules ---dhcpshow

Show LAN port IP address	---ipshow
Show NAT info	---natshow
Change LAN port IP address	---ip –addr x.x.x.x –mask x.x.x.x

Example:<config-interface-fastethernet-lan>#ip –addr 192.168.1.10 –mask 255.255.255.0

- **WAN interface settings**

path: <config-interface-fastethernet-wan>#

[disable]enable dhcp client	---[no]dhcp
[disable]enable pppoe	---[no]pppoe
[disable]enable QOS	---[no]qos
Set default gateway IP	---gateway x.x.x.x
Clear default gateway IP	---no gateway
Set WAN port IP address	---ip –address x.x.x.x -mask x.x.x.x

Example:<config-interface-fastethernet-wan>#ip –addr 202.112.241.100 –mask 255.255.255.0

You need to reconnect if the WAN port has been changed.

Show WAN port settings	---show

- **MMI Filter**

path: <config-mmifilter>#

add filter rule	---entry –start x.x.x.x –end x.x.x.x

Example:<config-mmifilter>#entry –start 202.112.20.1 –end 202.112.20.255

Delete filter rule	---no entry –start x.x.x.x

Example:<config-mmifilter>#no entry –start 202.112.20.1

Show filter rule	---show
[disable]enable MMI filter	---[no]start-filter

- **NAT settings**

path: <config-nat>#

[disable]enable ftp alg	---[no]ftpalg
[disable]enable ipsec alg	---[no]ipsecalg
[disable]enable pptp alg	---[no]pptpalg
Add TCP mapping rule	---tcp-entry –ip x.x.x.x –lanport xxx –wanport xxx

Example:<config-nat>#tcp-entry –ip 192.168.1.5 –lanport 1720 –wanport 1000

Delete TCP mapping rule ---no entry –ip x.x.x.x –lanport xxx –wanport xxx

Example:<config-nat>#no tcp-entry –ip 192.168.1.5 –lanport 5060 –wanport 1000

Add UDP mapping rule ---udp-entry –ip x.x.x.x –lanport xxx –wanport xxx

Delete UDP mapping rule ---no udp-entry –ip x.x.x.x –lanport xxx –wanport xxx

Show NAT info ---show

- **Netservice**

path: <config-netservice>#

Set DNS address ---dns -ip x.x.x.x _domain xxx

Example:<config-netservice>#dns –ip 202.112.10.36 _domain voip.com

Set alternate DNS address ---alterdns -ip x.x.x.x _domain xxx

Set hostname ---hostname xxx

Set http access port ---http-port xxx

Show http access setting ---http-port

Set telnet access port ---telnet-port xxx

Show telnet access port ---telnet-port

Set RTP initial port and quantity ---media-port –startport xxx –number xxxx

Example:<config-netservice>#media-port –startport 10000 –number 200

Add route rule ---route –gateway x.x.x.x –addr x.x.x.x –mask x.x.x.x

Example: Arcihfone<config-netservice>#route –gateway 202.112.10.1 –addr 202.112.210.1 –mask 255.255.255.0

Delete route rule ---no route –gateway x.x.x.x –addr x.x.x.x –mask x.x.x.x

Show route info ---route

Show netservice info ---show

- **Dial-peer settings**

path: <config-pbook>#

[disable]enable calling through GK and proxy ---[no]enableGKandProxy

Add number-IP bond entry ---entry –number xxx –ip x.x.x.x –protocol xxx

Example:<config-pbook>#entry –number 100 –ip 202.112.20.100 –protocol sip

Add number-IP bond and add prefix to the dial number

---entry –number xxx –ip x.x.x.x –protocol xxx _add xxx

Example:<config-pbook>#entry –number 100 –ip 202.112.20.100 –protocol sip _add 123(dial 100 and will send 123100 according this rule)

Add number-IP bond and replace the destination with another number

---entry –number xxx –ip x.x.x.x –protocol xxx _all xxx

Example:<config-pbook>#entry –number 100 –ip 202.112.20.100 –protocol sip _all 123(user dial 100 and gateway will sent 100 instead)

Add number-IP bond and delete the prefix of the destination number

---entry –number xxx –ip x.x.x.x –protocol xxx _del xxx

Example:<config-pbook>#entry –number 1234 –ip 202.112.20.100 –protocol sip _del 2 (dial 1234 will send 34 instead)

Add number-IP bond and replace the prefix with another number

---entry –number xxx –ip x.x.x.x –protocol xxx _rep xxx _length xxx

Example:<config-pbook>#entry –number 1234 –ip 202.112.20.100 –protocol sip _rep 567 _length 2(dial 1234 will send 56734)

Delete dial-peer entry	---no entry –number xxx
Show current dial-peer rules	---show
Set default voip protocol	---default-protocol xxx

- **Port settings**

path: <config-port># 或<config-port X>#

set accecp relay mode	---accept-relay xxx
set callerid mode	---callerid xxx
disable callerid	---no callerid

config call forward ---callforward –conditon xxx –number xxx –ip xxx –port xxx –protocol xxx

Example:<config-port 0>#callforward –condition busy –number 100 –ip 202.112.10.100 -port 5060 –protocol sip

Disable call forward	---no callforward
[disable]enable call transfer	---[no]calltransfer
[disable]enable call waiting	---[no]callwaiting

Set prefer codec ---codec xxx

Set DTMF gain ---dtmfvolume xxx

Set black list ---in-limit xxx

Show black list ---in-limit

Set input volume ---input xxx

Set outgoing limit list ---out-limit xxx

Show outgoing limit list ---out-limit

Set output volume ---output xxx

[disable]enable outgoing limit ---[no]shutdown out

[disable]enable black list ---[no]shutdown in

[disable]enable outgoing limit and black list ---[no]shutdown

[disable]enable 3-way conference ---[no]threetalk

Show port settings ---show

- **PPPoE settings**

path: <config-pppoe>#

PPPoE account settings ---auth –user xxx -password xxx

Example:<config-pppoe>#auth –user aaa –password 123456

[disable]enable service settings ---[no]service xxx

Show pppoe settings ---show

- **QOS settings**

path: <config-qos>#

[delete]add QoS table entry --- [no]entry –addr x.x.x.x –mask x.x.x.x

Example:<config-qos>#entry –addr 202.112.10.1 –mask 255.255.255.0

[disable]enable include QOS table ---[no]include

Show QoS settings ---show

- **SIP settings**

path: <config-sip>#

[disable]enable registration ---[no] register

[disable]enable auto detect server ---[no] detect-server

Set sip domain ---default-domain xxx

Set DTMF mode ---dtmf-mode xxx

Set auto detect interval time ---interval-time xxx

Set RFC edition ---rfc-version xxx

[disable]enable auto swap server --- [no]swap-server

Set sip account ---number-password –number xxx –password xxx

Set local SIP signal port --- signalport xxx

Set proxy server ---server proxy -ip x.x.x.x _port xxx _user xxx _password xxx

Example:<config-sip-server># proxy ip 210.25.23.22 _port 5060 _user aaa _password 123456

Set register server info ---server register -ip x.x.x.x _port xxx –user xxx _password xxx

Set alter proxy info ---alter-server proxy –ip x.x.x.x _port xxx _user xxx _password xxx

Set alter server info ---alter-server register –ip x.x.x.x _port xxx _user xxx _password xxx

[disable]enable stun server ---stun [no]enable

Set stun detecting interval time ---stun interval-time xxx

Set stun server ip and port ---stun –ip x.x.x.x –port xxx

Show current sip info ---show

- **User management**

path: <config-user>#

Change user right. ---access –user xxx –access xxx

Example:<config-user>#access –user aaa –access 7

Change user password ---password –user xxx

Add new user ---entry –user xxx –access xxx

Example:<config-user>#entry –user abc –access 7

Delete user entry ---no entry –user xxx

Show current sip info ---show

Debug (Level 0~7)

path: <debug>#

show debug setting ---show

[disable]enable debug all modules ---[no] all xxx

[disable]enable debug app module ---[no] app xxx

[disable]enable debug cdr module ---[no] cdr xxx

[disable]enable debug sip module ---[no] sip xxx

[disable]enable debug h323 module ---[no] h323 xxx

[disable]enable debug tel module ---[no] tel xxx

[disable]enable debug dsp module ---[no] dsp xxx

Download configure to flash

usage: #download tftp –ip x.x.x.x –file xxx

#download ftp –user xxx –password xxx –ip x.x.x.x –file xxx

Example: #download ftp –user abc –password 123 –ip 202.112.20.15 –file AG188.cfg

Password

usage: #password

Enter new password:xxx

Confirm new password:xxx

Reload

usage: #reload

Reboot system

Show system running info

- accesslist

path: <show>#

show: accesslist (firewall) settings

Example: #<show>#accesslist

- basic

path: <show>#

show network status

Example: #<show>#basic

- call

path: <show>#

show current call info

Example: #<show>#call active

- capability

path: <show>#

show CODEC capability

Example: #<show>#capability

- debugging

path: <show>#

show debug info

Example:#<show>#debugging

- dhcp-server

path: <show>#

show LAN status and DHCP server info

Example:#<show># dhcp-server

- dial-rule

path: <show>#

show digital-map info

Example:#<show># dial-rule

- interface

path: <show>#

show LAN info

Example:#<show>#interface fastethernet lan

show WAN info

Example:#<show>#interface fastethernet wan

- ip

path: <show>#

show arp table info

Example:#<show>#ip arp

Show DNS server info

Example:#<show>#ip dns

Show netstate info

Example:#<show>#ip netstat

Show route info

Example:#<show>#ip route

Show icmp packets Stat.

Example:#<show>#ip icmp

Show igmp packets Stat.

Example:#<show>#ip igmp

Show ip packets Stat.

Example:#<show>#ip ip

Show RTP packets Stat.

Example:#<show>#ip rtp

Show TCP packets Stat.

Example:#<show>#ip tcp

Show UDP packets Stat.

Example:#<show>#ip udp

- memory

path: <show>#

show IP phone memory

Example:#<show>#memory

- nat

path: <show>#

show NAT information

Example:#<show>#nat

- port

path: <show>#

show caller-ID info

Example:#<show>#port callerID

show dsp info

Example:#<show>#port dsp

show hotline info

Example:#<show>#port hotline

show black list info

Example:#<show>#port in-limit

show outgoing limit info

Example:#<show>#port out-limit

show current phone number

Example:#<show>#port number

show current port status

Example:#<show>#port status

- PPPoE

path: <show>#

show PPPoE info

Example:#<show># pppoe

- qos

path: <show>#

show QoS table info

Example:#<show>#qos

- sip

path: <show>#

show sip info

Example:#<show>#sip

- udptunnel

path: <show>#

show UDP tunnel info

Example:#<show># udptunnel

- uptime

path: <show>#

show running time

Example:#<show># uptime

- version

path: <show>#

show IP phone version

Example:#<show># version

Telnet and logout

Usage: #telnet –target -port

Login:xxx

Password:xxx

#

#logout

Telnet and logout

path: <time>#

---manualset –year xxx –month xxx –day xxx –hour xxx –minute xxx –second xxx

Example:<time>#manulset –year 2004 –month 10 –day 1 –hour 8 –minitute 30 –second 0

[disable]enable SNTP server	---sntp [no] start
Set SNTP IP address	---sntp server x.x.x.x
Set SNTP server timeout	---sntp timeout xxx
Set timezone (-12~+12)	---sntp zone xxx
Show SNTP info	---sntp show
Show current time	---print

Tracert trace network path info

usage: #tracert –host

Example:#tracert 3 HYPERLINK "http://www.google.com" 4www.google.com5

Update IP Phone

usage: # update ftp –user xxx –password xxx –ip x.x.x.x –file xxx

update tftp –ip x.x.x.x –file xxx

Example:# update ftp –user abc –password 123 –ip 202.112.20.15 –file AG188.dlf

Upload configure file

usage: # upload ftp –user xxx –password xxx –ip x.x.x.x –file xxx

upload tftp –ip x.x.x.x –file xxx

Network Diagnosis

There are some telnet commands for checking your network. Now Listing below for your information

Command	Function	Example

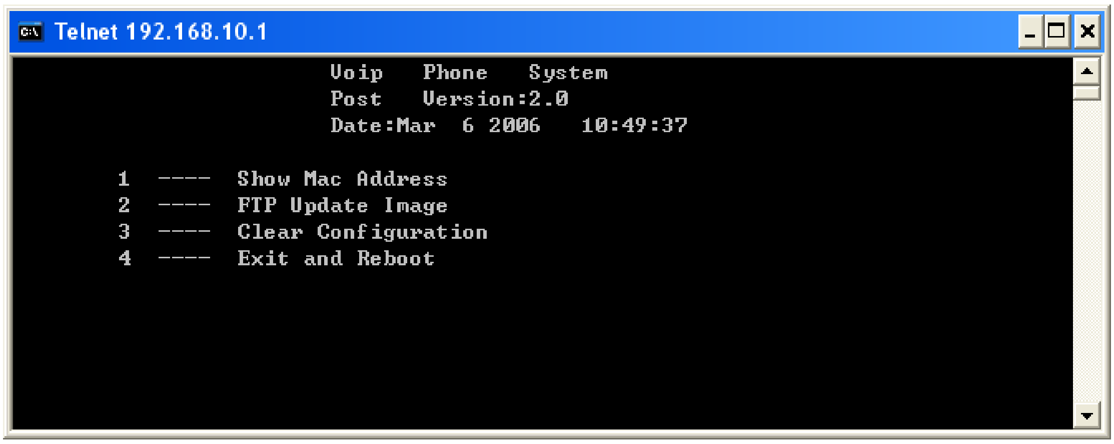
ping	Check if the destination is accessible	#ping www.google.com
tracert	Show network path info	#tracert www.google.com
show basic	Show network settings	#show basic
show ip route	Show route table	#show ip route
show ip arp	Show arp table	#show ip arp
show ip netstat	Netstat programe	#show ip netstat
telnet	Telnet to another device	#telnet 192.168.1.2

Reset to factory default

#setdefault clear IP phone settings expect network part

#setdefault all clear all settings.

POTS Mode (Safe mode)



VIP-155PT provide safe mode. When there is booting problem because of setting problem or firmware problem. User can restore the factory setting or upgrade to a new firmware to solve this problem.

How to enter safe mode?

There will be a schedule bar in the VIP-155PT booting procedure, press # key within the first 5 seconds, then the phone will go to POST mode. It has a default ip 192.168.10.1 in POST mode. User may change the PC's IP address to 192.168.10.xx and telnet to 192.168.10.1 to access the VIP-155PT in POST mode.

User can accord the guide in post mode to clear the settings or upgrade the firmware.

Appendix A

FAQ

Q1: How many servers may VIP-155PT register simultaneously?
A1: VIP-155PT is able to register two SIP servers simultaneously, and redundancy servers. User can configure the dial peer to route calls between these servers.
Q2: Why the settings vanish after reboot?
A2: Please go to Config Manage→Save Config to save your setting always.
Q3: How to use speed dial function?
A3: There are 9 speed dial keys in the IP PHONE panel, Usage: Set speed dial number: press the speed key and enter the speed dial number and then press Menu/OK key to save the setting. Pick up the handset and press the speed dial key to dial the pre-define number.
Q4: How to use set the IP type via keypad?
A4: In the idle mode, user may us the keypad to set the IP type as the below procedure: Keep pressing the button 1 for changing to static mode. Keep pressing the button 2 for changing to DHCP mode. Keep pressing the button 3 for changing to PPPoE mode.

Appendix B

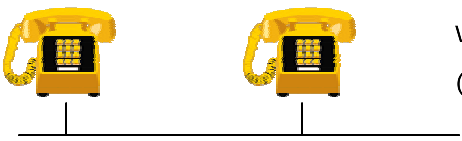
Voice communications

There are several ways to make calls to desired destination in IP Phone. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

Peer to Peer (P2P) Mode

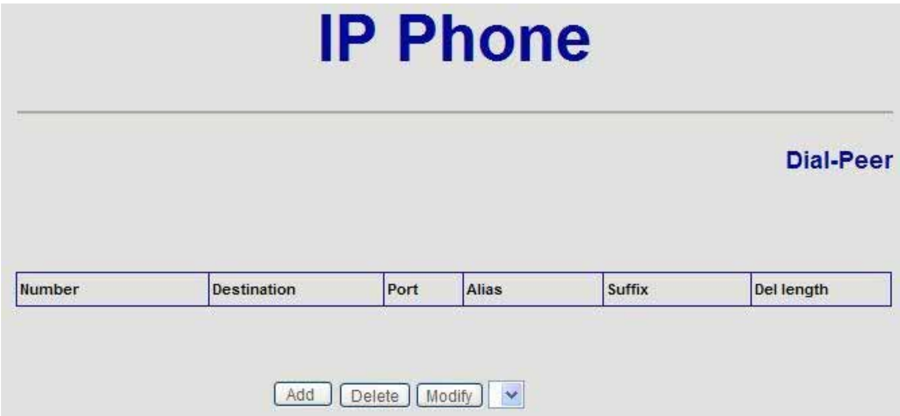
Step 1: Assuming there are two VIP-155PT in the network the IP address are 172.16.0.1 and 172.16.0.2

VIP-155PT-A
WAN IP Address
(172.16.0.1)

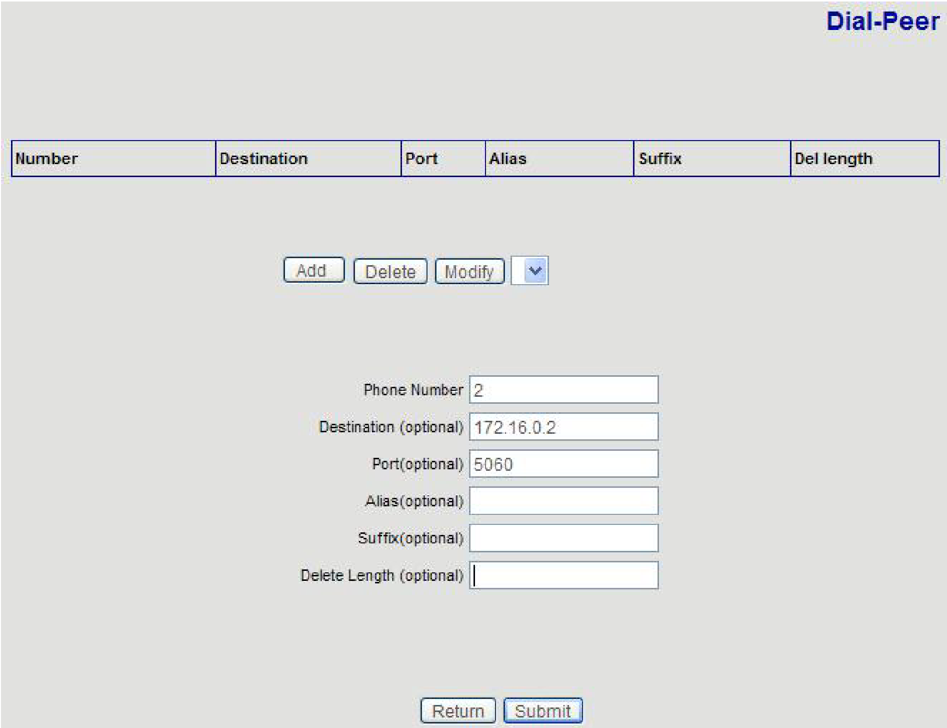


VIP-155T-B
WAN IP Address
(172.16.0.2)

Step 2: Execute your web browser, and insert the IP address (**172.16.0.1**) of the VIP-155PT-A in the adddress bar. After log on machine, browse to "**Dial-peer**" configuration item:



Step 3: Press "Add" button and fill in the below parameter, be sure to click the "**Submit**" button to apply settings. Browsing to "**Config Manage**" → "**Save Config**" configuration item and press "**Save**" button to save the configuration.

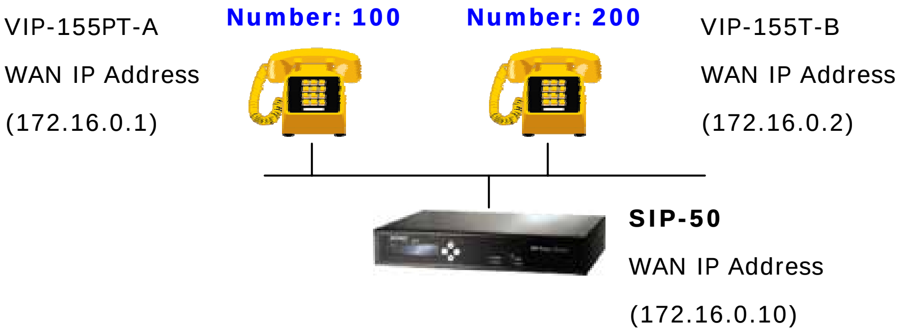


Step 4: Pick up handset or press "Handfree" key from keypad of VIP-155PT-A and dial "2#". Then the phone of VIP-155PT-B should ring. You can do the same thing to the VIP-155PT-B.

ⓘ Hint

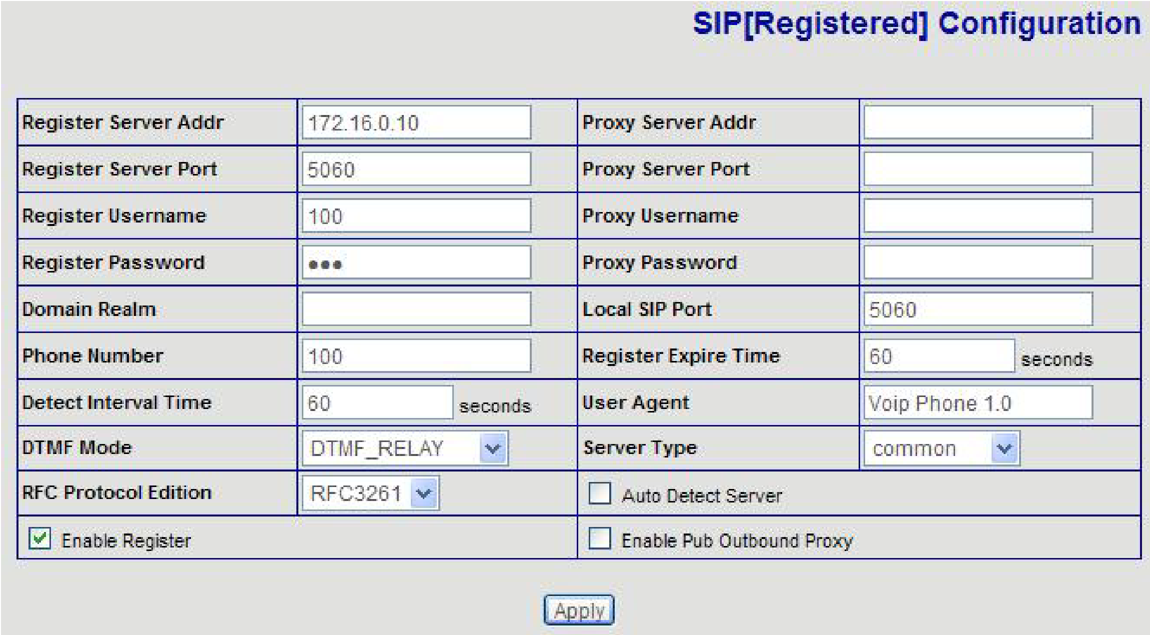
- If the IP address of the remote calling party is known, you may directly make calls by preset number via its IP address and end with an "#".
- If the IP phones are installed behind a NAT/firewall/ IP sharing device, please make sure the NAT device support SIP applications before making calls

Proxy Mode



SETP 1:

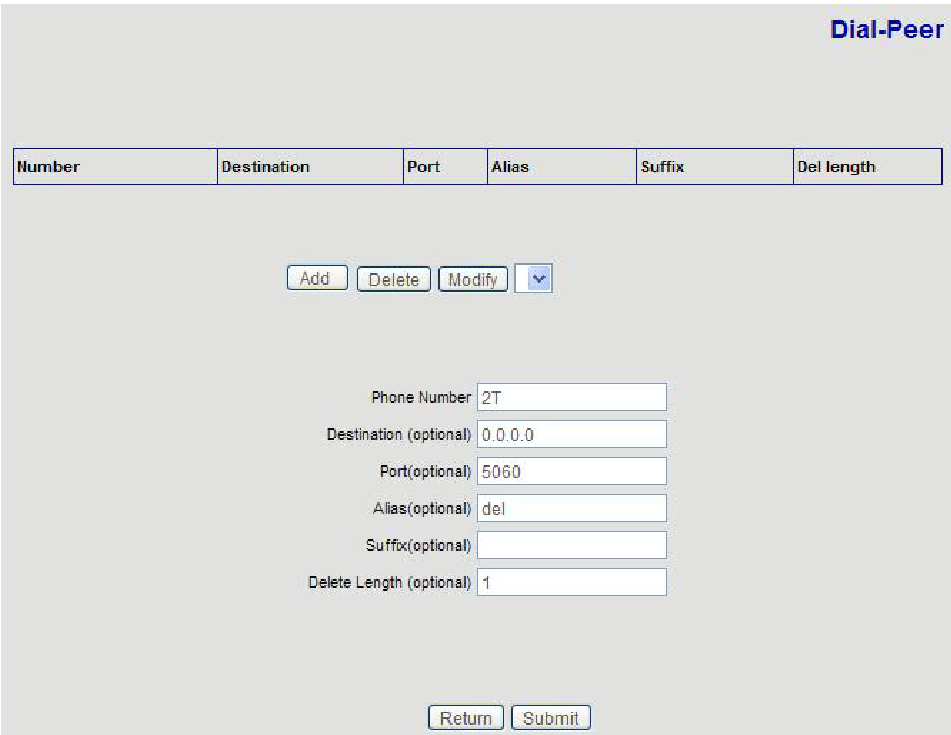
Please browse machine "**VoIP**" → "**SIP Config**" menu, and enable the "**Enable Register**" check box. Insert IP address of the remote calling party in the "**Register Server Addr**" field. Sample configuration screen is shown below:



After these configurations, be sure to click the "**Apply**" button to apply settings.

SETP 2:

Browsing to "**Dial-peer**" configuration item, press "**Add**" button and fill in the below parameter.



After these configurations, be sure to click the "**Apply**" button to apply settings.

SETP 3:

Browsing to "**Config Manage**" → "**Save Config**" configuration item and press "**Save**" button to save the configuration. Browsing to "**System Manage**" → "**Reboot**" menu and press "**Reboot**" button reboot the machine to make the settings effective. After rebooting, the unit will register to SIP-50, the LCD screen will show below:

```
VOIP PHONE
   SEP 20 13 12:30
```

SETP 4:

At this moment, you may pick up the handset and dial "200" to connect with extension 200 of VIP-155PT-B to start the voice communications.

Appendix C

VIP-155PT series Specifications

Product	Power over Ethernet SIP IP Phone
Model	VIP-155PT
Hardware	
WAN	1 x 10/100Mbps RJ-45 port Power Over Ethernet 802.3af compliant at PT model
LAN	1 x 10/100Mbps RJ-45 port
LCD display	2 x 16 characters
Speaker	8 Ohm/0.2 Watt speaker for speakerphone operation
Protocols and Standard	
Standard	SIP 2.0 (RFC3261), SIP digest authentication (MD5)
Voice codec	G.723.1 (6.3k/5.3k), G.729, G.711 (a-law/u-law)
NAT Traversal	Outbound Proxy, STUN
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Dynamic Jitter Buffer
Supplementary services	Immediate (unconditional) call forwarding Busy call forwarding No answer calls forwarding Calls hold/transferring. Answer Machine 3-Way conference calls
Call history	Incoming call Outgoing call Missed (not accepted) call history Voice Record
Protocols	TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, FTP, PPP, PPPoE
Network and Configuration	
Access Mode	Static IP, PPPoE, DHCP
Management	Web, Keypad, Telnet
Dimension (W x D x H)	200 mm x 184 mm x 60 mm
Operating Environment	0~40 degree C, 10~90% humidity
Power Requirement	12V DC (802.3af 48VDC in line power)
EMC/EMI	CE, FCC Class B